NCS TIB 81-4

NATIONAL COMMUNICATIONS SYSTEM

TECHNICAL INFORMATION BULLETIN.

(14) NCS-TIB-81-4

(6)

CCITT STUDY GROUP XVIII WORK PROGRAM 1981-84; (INTEGRATED SERVICES DIGITAL NETWORK)

(11) JUNE 1981 (12) 179

APPROVED FOR PUBLIC RELEASE
DISTRIBUTION UNLIMITED

Unclassified

SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered)

REPORT DOCUMENTATION PAGE

READ INSTRUCTIONS BEFORE COMPLETING FORM

1. REPORT NUMBER
NCS TIB 81-4

2. GOVT ACCESSION NO.

3. RECIPIENT'S CATALOG NUMBER

4. TITLE (and Subtitle)
1981-84 CCITT Study Group XVIII Work Program (Integrated Services Digital Network (ISDN))

5. TYPE OF REPORT & PERIOD COVERED
Interim

6. PERFORMING ORG. REPORT NUMBER

7. AUTHOR(s)
International Telegraph and Telephone Consultative Committee (CCITT)

8. CONTRACT OR GRANT NUMBER(s)

9. PERFORMING ORGANIZATION NAME AND ADDRESS

10. PROGRAM ELEMENT, PROJECT, TASK AREA & WORK UNIT NUMBERS

11. CONTROLLING OFFICE NAME AND ADDRESS

12. REPORT DATE
June 1981

13. NUMBER OF PAGES
179

14. MONITORING AGENCY NAME & ADDRESS (If different from Controlling Office)
Office of Technology and Standards
National Communications System
Washington, D.C. 20305

15. SECURITY CLASS. (of this report)
Unclassified

15a. DECLASSIFICATION/DOWNGRADING SCHEDULE

16. DISTRIBUTION STATEMENT (of this Report)
Distribution unlimited; approved for public release.

17. DISTRIBUTION STATEMENT (of the abstract entered in Block 20, if different from Report)

18. SUPPLEMENTARY NOTES

19. KEY WORDS (Continue on reverse side if necessary and identify by block number)
ISDN
CCITT Study Group XVIII
Integrated Digital Networks
Timing
Speech Coding

20. ABSTRACT (Continue on reverse side if necessary and identify by block number)
This Technical Information Bulletin (TIB) preprints the text of the Questions, approved by the International Telegraph and Telephone Consultative Committee (CCITT) 1980 Plenary Assembly, for study by CCITT Study Group XVIII during the 1981-84 study period. These questions, relating to the Integrated Services Digital Network (ISDN), provide an insight into the direction international standards and networks are heading, and how they will look in the future.

DD FORM 1 JAN 73 1473 EDITION OF 1 NOV 65 IS OBSOLETE

Unclassified

SECURITY CLASSIFICATION OF THIS PAGE (When Data Entered)

NCS TECHNICAL INFORMATION BULLETIN 81-4

2

1981-84 CCITT STUDY GROUP XVIII WORK
PROGRAM (INTEGRATED SERVICES DIGITAL NETWORK (ISDN))

JUNE 1981

PROJECT OFFICER:

ROBERT M. FENICHEL
Electronics Engineer
Office of NCS Technology
and Standards

APPROVED FOR PUBLICATION:

Marshall L. Cain
MARSHALL L. CAIN
Assistant Manager for
NCS Technology and
Standards

FOREWORD

Among the responsibilities assigned to the Office of Technology and Standards, National Communications System (NCS), is the management of the Federal Telecommunication Standards Program, which is an element of the overall GSA Federal Standardization Program. Under this program, the NCS, with the assistance of the Federal Telecommunication Standards Committee, identifies, develops, and coordinates proposed Federal Standards which either contribute to the inteoperability of functionally similar Federal telecommunication networks or to the achievement of a compatible and efficient interface between computers and telecommunications. In developing and coordinating these standards, considerable effort is expended in initiating and pursuing joint standards development efforts with appropriate technical committees of the Electronic Industries Association, the American National Standards Institute, the International Organization for Standardization, and the International Telegraph and Telephone Consultative Committee (CCITT) of the International Telecommunications Union. This Technical Information Bulletin presents a reprint of questions allocated to CCITT Study Group XVIII for the 1981-1984 plenary period (document COM XVIII - No. 1-E). These questions, relating to the Integrated Services Digital Network (ISDN), provide an insight into the direction international standards and networks are heading, and how they will look in the future. Any comments, inputs or statements of requirements that could assist in the advancement of this work are welcome and should be addressed to:

Office of Technology and Standards
National Communications System
Washington, DC 20305
Telephone (202) 692-2124

TABLE OF CONTENTS

International Telegraph and Telephone
Consultative Committee

(CCITT)

Period 1981-1984

COM XVIII-No. 1-E

Original : English

Questions : 1 to 19/XVIII

Date : February 1981

STUDY GROUP XVIII - CONTRIBUTION No. 1

SOURCE : VIIth PLENARY ASSEMBLY OF THE CCITT

TITLE : QUESTIONS ALLOCATED TO STUDY GROUP XVIII FOR THE PERIOD 1981-1984

1. This contribution gives the texts of the Questions approved by the VIIth Plenary Assembly of the CCITT for study by Study Group XVIII in the period 1981-1984.

Administrations, recognized private operating agencies, international organizations and scientific or industrial organizations which intend to take part in the work of Study Group XVIII are invited to prepare contributions to the study of these Questions and to send them to the CCITT Secretariat as soon as possible.

2. The attention of participants in the work of the CCITT is drawn to the following provisions extracted from the "Additional rules of procedure of the CCITT" (as amended by decisions of the VIIth Plenary Assembly, to appear in the Yellow Book, Volume I, Resolution No. 1) :

2.1 General

i) "Contributions which are to be distributed in the normal way before the meeting in the three working languages of the Union (as white documents) must reach the CCITT Secretariat at least three months before the date of the meeting for which they are intended."

ii) "If there are insufficient contributions, no meeting at all should be scheduled."

2.2 Form of submission of contributions

"Contributions should be presented in accordance with the following general directives :

i) Contributions should be concisely drafted, avoiding any unnecessary details, tables or statistics that make no direct contribution to the study of a Question. They should be clearly written with a view to being universally understood, that is, as codified as possible, using international terminology and avoiding the technical jargon peculiar to the author's country. When a contribution deals with several Questions, the Questions should be separated so that the text of each Question begins on a fresh sheet of paper (not on the back of a page).

ii) A contribution should not as a general rule exceed about 2,500 words (five pages), nor include more than three pages of figures (making eight pages in all). The contribution should be accompanied by a summary and should be followed by conclusions whenever possible. For draft Recommendations and for contributions submitted by Special Rapporteurs, the above directives should not apply.

iii) Documents of purely theoretical interest which are not directly related to the Question under study should not be submitted in their entirety. Short abstracts only of such articles could be sent to the CCITT for translation and publication.

iv) Articles which have been or will be published in the technical press should not be submitted to the CCITT. Short abstracts only of such articles could, however, be sent to the CCITT for translation and publication.

v) Passages of an unduly commercial nature included in a contribution may be deleted by the Director of the CCITT in agreement with the Chairman; the author of the contribution will be advised of any such deletions."

2.3 Distribution and format

i) "Contributions should be drafted in one (or more) of the working languages of the Union and three copies sent to the CCITT Secretariat, with further copies to be sent by contributors direct to the Chairman and Vice-Chairmen of the Study Group and to the relevant Chairmen of the Working Parties and Special Rapporteurs.

It is recommended that a translation of contributions in another working language be sent to the CCITT Secretariat."

ii) "Contributions should be submitted on very white paper of A4 format, in clear back type. If the paper is not of A4 format, the text on each page should not exceed that format.

The top half of the first page should be left blank."

3. The addresses of the Chairman and Vice-Chairmen of Study Group XVIII are given in Table 1.

TABLE 1

Study Group XVIII - List of addresses

Name	Address	Telephone	Telex
T. IRMER (Chairman)	Abteilungspräsident, A3-PO Box 5000 Fernmeldetechnisches Zentralamt Am Kavalleriesand 3 D-6100 DARMSTADT (Federal Republic of Germany)	+49 6151 83 1 or +49 6165 834800	419511 ftzd
M. DECINA (Vice-Chairman)	S.I.P. - D6 Casella postale 2420 1-00100 ROMA A.D.R.M. Italy	+39 636 883	61 467 sipgen
V.I. JOHANNES (Vice-Chairman)	Bell Telephone Laboratories HOLMDEL, N.J. 07733 United States	201-949-6391	132450
K. OKIMI (Vice-Chairman)	Engineering Bureau NTT 1-1-6 Uchisaiwaicho Chiyoda-Ku TOKYO 100 Japan	083-509-5335	14970
H.K. PFYFFER (Vice-Chairman)	Direction générale des PTT Chief of Section "Transmission" TZV 3030 BERNE Switzerland	41 031 62 1111	32061 vptt ch
B. ROCHE (Vice-Chairman)	PTT Centre National S'Etudes des Télécommunications 22300 LANNION France	33 96 38 11 11	

4. List of Questions assigned to Study Group XVIII for the period 1981-1984.

TABLE 2

Questions	Title	Remarks
1/XVIII *)	General network aspects of an Integrated Services Digital Network (ISDN)	See Annex **)
2/XVIII	Customer/network interface	
3/XVIII	Synchronization in digital networks	
4/XVIII *)	Signalling for the ISDN	
5/XVIII *)	Switching for the ISDN	
6/XVIII	Definition for digital networks	
7/XVIII	Encoding of speech and voice-band signals using methods other than PCM, in accordance with Recommendation No. G.711	
8/XVIII	Digital speech interpolation system	
9/XVIII *)	General network performance aspects of integrated digital networks	
10/XVIII *)	Availability for the ISDN	
11/XVIII	Characteristics for digital sections	

*) Urgent Questions. Also Questions 3, 6, 12, 13 and 15/XVIII could be considered as urgent Questions.

**) The Annex to the list of Questions deals with the study of the Integrated Services Digital Network (ISDN).

TABLE 2 (Cont'd.)

Questions	Title	Remarks
12/XVIII	Maintenance philosophy of the digital network	Of interest to Study Group IV **)
13/XVIII	Implementation of maintenance philosophy	
14/XVIII *)	Interworking between digital systems based on different standards	
15/XVIII	Interfaces in digital networks	
16/XVIII	Performance characteristics of PCM channels at audio frequencies	
17/XVIII	Characteristics of PCM multiplexing equipment and other terminal equipments for voice frequencies	
18/XVIII	Characteristics of digital multiplex equipment and multiplexing arrangements for telephony and other signals	
19/XVIII	Network aspects of existing and new levels in the digital hierarchy	

*) Urgent Questions. Also Questions 3, 6, 12, 13 and 15/XVIII could be considered as urgent Questions.

**) The Annex to the list of Questions deals with the study of the Integrated Services Digital Network (ISDN)

QUESTION 1/XVIII - General network aspects of an Integrated Services Digital Network (ISDN) (continuation of part of Question 1/XVIII, studied in 1977-1980)

This Question is concerned with overall studies related to the general features of future Integrated Services Digital Networks capable of satisfying the requirements of many different services. Study Group XVIII will define the scope and framework of an ISDN and identify the services which may be incorporated in such networks. It will study the evolution of Integrated Digital Networks (IDNs) dedicated to specific services (e.g., telephony, data) towards an ISDN.

The objectives will be to define overall network and system principles which can form a basis for study and Recommendations by appropriate specialist CCITT Study Groups. The generic features appropriate and applicable to an ISDN will be identified together with optional service dependent features applicable to part of an ISDN.

The study of the following five related aspects will take into account the considerata arising from studies carried out during the 1977-1980 study period as recorded in Annex A to this Question. In addition, the multiple aspects of this work require coordination between the various Study Groups involved (e.g., Study Groups III, VII, XI, XV, XVI, XVII and XV).

Some of these Questions have to be studied initially by Study Group XVIII, with high priority, to enable other Study Groups to initiate or continue their work and to draft Recommendations within the current CCITT study period. In other cases Study Group XVIII needs information from other Study Groups in order to make progress in its own network studies.

Recommendation No. G.705 provides information and future developments of the ISDN.

Studies of ISDN aspects were carried out under Question 1/XVIII during the 1977-1980 study period and a partial reply to that Question is reproduced as Annex 1 to this new Question. Annex 2 records many points already identified and of relevance to the ongoing studies. Annexes 3 and 4 contain significant information which was not fully considered before the end of the study period. These Annexes are also of relevance to other new Questions of Study Group XVIII.

Note : The Chairmen and Vice-Chairmen of the Study Groups involved (Study Groups III, VII, XI, XV, XVI, XVII and XVIII) will jointly assess the progress made by the various Study Groups and initiate any steps necessary to expedite the work. This should take place at about the middle of the study period (e.g. beginning of 1982), with the Chairman of Study Group XVIII acting as convenor for this coordination.

Considering

a) that the requirements of data transmission services and several new non-voice services are being studied by CCITT.

Note : In several countries services dedicated digital networks are already in service or will be installed for non-voice services that may use part of the ISDN for access to this network.

b) many countries wish to adopt a common strategy for extending the use of Integrated Digital Networks (IDN) beyond the telephony application to form Integrated Services Digital Networks,

c) telephony service will constitute the major portion of the carried load on digital networks characterized by time division transmission and switching and common-channel signalling,

d) efficiency and economy of methods of access to the ISDN from customer terminals are significant factors in planning the local network,

e) CCITT Recommendations on digital switching and inter-exchange signalling, which take into account the future evolution of the IDN for telephony towards the ISDN, are already available in the Q series and may form the basis for further Recommendations on ISDN.

Point A. Service aspects

1. Which services should be taken into account in the establishment of network features of the ISDN ?

2. What are the network features needed to support these services ? Which network features should be regarded as general throughout the ISDN, and which should be classed as service dependent for particular service applications ?

Note : Among other network features, attention should be paid to charging so that adequate information could be made available for charging purposes.

3. For which services, if any, should a change of service on an established connection be envisaged ? What are the implications and requirements of such a feature ?

4. What kinds of leased paths will be required in the ISDN when it is in widespread operation ?

Note 1 : Services should be identified which will supplant existing leased line services.

Note 2 : Consideration should be given to the use of semi-permanent connections, closed user group and hot-line features, remote switching units etc.

Point B. Network aspects

1. What are the principles in terms of network structure and systems architecture which define the ISDN and which form the basis for study of specific aspects ?

2. Should layered protocols and functional layers be adopted for ISDN to form the basis of CCITT Recommendations ? If so, what are the characteristics of this layering, and in which way is the concept of functional layers used with respect to sub-systems, such as, e.g., the signalling channels ?

3. What are the implications of ISDN on numbering plans and service indicators for telephony and other services ?

4. What methods of voice band encoding other than standard PCM (see also Question 7/XVIII) and what forms of digital speech interpolation can be considered in relation to the evolution of the ISDN ?

Point C. Customer access

What are the principles in terms of network structure and systems architecture which define customer access to ISDN and which should form the basis of studies of related transmission, switching, signalling and interface aspects ?

Point D. Interworking

What are the principles which should form the basis for detailed study of the interfaces interconnections and interworking between ISDN and service dedicated networks ?

The following specific points should be inc[illegible] in the studies :

i) At what point in the connection should spe[illegible] [illegible]cessing for interworking be accomplished (e.g., in the originating o[illegible] [illegible]ter[illegible]inating country) ?

ii) What networks should be given preference to complete connections in a transit call situation ?

iii) What special problems arise from the use of ISDN to provide interconnections of particular services (e.g., according to X.21, etc.) via different networks, and what restrictions or restraints should be placed on services or networks when interworking (e.g., to accommodate accounting, timing and signalling, features) ?

iv) What methods should be recommended for acc[illegible]sing one network from another ?

v) How should conversions be accomplished (e.g. data to data, voice to data) ?

vi) What arrangements or procedures are needed to accommodate the accounting function for a connection involving mixed networks ?

vii) What influence would different national applications of service integration have on the international network with regard to interworking ?

viii) What special problems arise from the use of ISDN to interconnect networks carrying services to existing standard terminal interfaces ?

ix) What are the possibilities of application of service bits allocated in primary PCM and higher order digital systems in national and international digital networks ?

Point E. Guidelines to facilitate evolution towards ISDN

Which strategy should be followed in order to facilitate and speed up the establishment of a worldwide ISDN ?

Note : It should be taken into consideration that, in the introductory period, it will be necessary to establish an all-digital network mainly for the needs of "business subscribers" who represent only a small percentage of the overall number of subscribers but who originate a substantial portion of the traffic. It may be useful to create a digital "overlay network" in each country and to interconnect these national networks by digital links.

Annex 1

(to Question 1/XVIII)

Partial reply to Question 1/XVIII, point A
(study period 1977-1980)

1. Introduction

During the 1977-1980 study period, members of Study Group XVIII have reflected increasing interest in ISDN as a possible means of enhancing telecommunication networks to support an increasing range of services. An important aspect of the work has been the extension of digital techniques to the customers' premises to give digital access to ISDN.

The relevant part of Question XVIII/1 is reproduced for reference purposes : "On what general philosophy should the design and introduction of digital systems be based ? For example, what principles should be applied for the implementation of dedicated integrated digital networks (IDNs) for various services and what provisions should be made to facilitate the evolution towards the possible future integrated services digital network (ISDN) ?"

2. Recommendation G.705 provides information concerning the future development and evolution of the ISDN.

3. In view of the urgent interest in general ISDN matters, and the availability of recent documentation which had not been fully discussed before the final meeting of Study Group XVIII, it was agreed that a means should be found of continuing the work and preparing documentation to form an early input to studies in the next study period. Although the formal CCITT organization does not make specific provisions to continue studies during the transition period from one plenary period to the next, Study Group XVIII invited the Rapporteur for ISDN aspects (Question 1/XVIII, point A of the 1977-1980 study period) to continue work by correspondence with delegates of other countries who had already expressed a wish to participate in this work. It was also foreseen that it might be desirable to have a meeting of those involved very early in the next study period. An approach would be made to the Chairman designate of Study Group XVIII to make the arrangements, should such a meeting be necessary.

4. In order to give some interim guidance to national studies of ISDN and related development work, Study Group XVIII drew attention to the points of view expressed in the following paragraphs. These were supported by Study Group XVIII and therefore reflect confidence in the approach indicated. The points are recorded under headings which identify different aspects of ISDN and facilitate separation of the subject into reasonable study areas.

5. Service aspects

Information exists on a wide range of existing and new voice and data services, and it is recognized that ISDN has generic features capable of supporting many of these. In addition there is scope for adding service-dependent features to appropriate parts of the network to satisfy particular requirements or to give interworking access to service-dedicated networks. Thus the possible implications of all services will need to be considered in the future, but it is proposed that initial attention should be concentrated on :

- Digital telephony to ensure that adequate provision is made for the predominant service.

- New services for which the capabilities available on ISDN are sufficient.

6. Network aspects

The telephone network will evolve towards an IDN with switched 64 kbit/s telephony channels and it is expected that other services would be integrated with that network during evolution towards an ISDN.

Constraints may be placed on the use of the 64 kbit/s capacity to accord with internationally agreed standards for some services. Connections through ISDN could be switched or semi-permanently connected and "wider-band" services may be carried by using multi-slot connections at n x 64 kbit/s. All ISDN exchanges are expected to have stored program control and inter-change signalling (CCITT No. 7 enhanced), with digital transmission paths on routes offering full ISDN service capabilities. Provision must be made for restricting service when interworking with equipment or networks having limited capability (e.g. calls routed through transmultiplexers or into the analogue telephony network).

7. Local network access

Digital transmission techniques in the local network will extend ISDN to customers' premises over a basic access which may, for example, be at 72 or 80 kbit/s. Other types of access will be considered as appropriate. The basic access will then provide a 64 kbit/s information channel and a separate channel. The 64 kbit/s channel may be dedicated to a particular service or used alternatively for voice or data, and on an established digital connection the path may be sub-divided for several lower-rate services. Capacity of the separate channel is expected to be 8 or 16 kbit/s and would carry customer/network signalling and possibly low speed telemetry.

Two methods of further exploiting the separate channel have been identified :

i) One method foresees that the separate channel would be dynamically allocated also to carry a form of data-message service.

ii) In another method the separate channel would be sub-multiplexed into two channels of Δ_1 and Δ_2 kbit/s. One of these channels would be used for data services at up to Δ_2 kbit/s in the local network, with rate adaptation for switching through ISDN at 64 kbit/s. The other at Δ_1 kbit/s would carry customer/network signalling for the 64 and Δ_2 kbit/s channels, and possibly low speed telemetry.

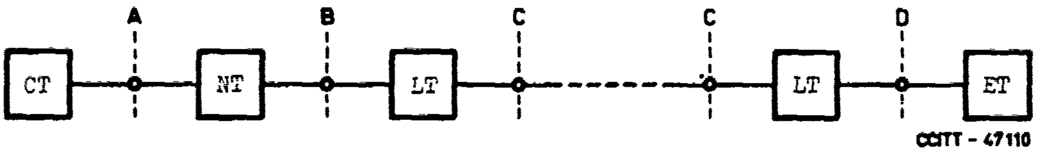
Alignment information for the basic access should also be provided, e.g. exploitation of the line transmission system; allocation of capacity within the 72 or 80 kbit/s.

Where justified (e.g. PABXs) a primary order digital path carrying a multiplex of 24 or 30 channels may be extended to a customers' premises, to provide several 64 kbit/s channels (see also paragraph 9). Customer/network signalling may be concentrated in one 64 kbit/s channel. This structure could also apply where a group of ISDN customers are connected via a multiplexer in the local network.

8. Customer interface

Figure 1 shows the functional elements of customer access to ISDN and interfaces A and B apply to the customers' premises. A considerable amount of further study is needed to establish preferred arrangements for connecting the wide range of voice and data terminals, some of which already exist with defined interface standards. Possible ISDN accesses at 72 or 80 kbit/s and 1,544 or 2,048 kbit/s would apply to interface B and little would be needed in the form of Network Termination for terminals which conform to these accesses. Connection of other terminals at interface A would require appropriate conversion functions in the Network Termination.

The layered model approach, devised for data services (see Study Group VII reference) may offer a convenient method of assisting the definition of the characteristics of interfaces A and B, which must be studied in conjunction with the customer/network signalling (paragraph 10).



CT - Customer Terminals

NT - Network Termination

LT - Line Termination

ET - Exchange Terminal

Figure 1 - Functional interfaces for digital local access

9. Local network transmission

Standards are available for digital transmission between exchanges and these could form a basis for transmission in the local network. Studies of ISDN access show a need for at least two types of system : one operating at current hierarchical rates (e.g. 1,544 or 2,048 kbit/s) and another to carry the basic access proposed in paragraph 7. Future studies may identify the need for other systems, including a smaller capacity multiplex for operating over existing local network cables. Specification of interfaces B and D will allow evolution of the transmission system somewhat independently of terminal and exchange equipment, in particular it is expected that new transmission media including optical fibre, coaxial and radio systems will be used as appropriate.

10. Customer/network signalling

Digital access to ISDN will include a separate channel to carry customer/ network signalling and possibly other information as described in paragraph 7. The Link Access protocols of the signalling system carried in this channel may be based on the exchange of frame formatted information using procedures similar to those recommended in level 2 of SS CCITT No. 7 and Recommendation X.25. This approach should give a very flexible, open-ended signalling capability, compatible with the requirements of new telecommunications terminals. The use of modern technology should ensure that the relative complexity of this method does not lead to excessive cost.

The Call Control aspects of the customer/network signalling require further study, taking into account information relating to existing telephony service (loop/disconnect and multi-frequency push-button systems) and circuit switched data services (X.21).

The customer/network signalling should also operate where a multiplexer is used in the local network. Signalling messages could, for example, be concentrated/distributed between several digital local lines and a common channel to the local exchange. This application would be very similar to the digital PABX served by a multiplex system and the studies of customer/network signalling should consider both situations.

11. <u>Switching aspects</u>

Digital trunk and local exchanges are already required to operate with digital transmission systems, and primary and secondary order multiplex interface standards for digital exchanges are available at 1,544, 2,048 and 8,448 kbit/s. To a large extent these standards could apply also when multiplex systems are used on the local network side of the exchange.

A new interface must be specified for the basic access described in paragraph 7, giving due consideration to the means of implementing conventional telephony functions (BORSCHT) and including any adaptation of other services for switching at 64 kbit/s. This interface may be similar to that existing at the local access to a remote multiplexer or concentrator.

12. <u>Interworking</u>

While it has been recognized that ISDN may be used as a means of access to service-dedicated networks the interworking arrangements have not yet been studied in detail. However, interworking in such cases is expected to be at the inter-exchange level, possible via nominated "gateway" nodes, and the arrangements could be based on specifications existing for inter-exchange signalling. (CCITT No. 7 as used in the telephony network, X.75 as used in Packet Data Networks and X.60 as used in Circuit Switched Data Networks.)

13. <u>Numbering and addressing</u>

Study Group XVIII recognizes the importance of the numbering and addressing arrangements where different services are carried on an integrated services digital network. Since the telephony based on IDN is expected to be an important basis for the development of the ISDN, the numbering scheme could similarly grow from that used for the telephony service. Further study is needed to establish a flexibile scheme, with a degree of independence between the identity of a network termination and the services available to a customer on that termination. The Study Group VII proposals for dedicated data networks have some relevance.

Annex 2

(to Question 1/XVIII)

A basis for study of ISDN

Studies of the arrangements for customer access to the ISDN through Contributions and discussions have lead to the establishment of some study points relating to system structure and network architecture. These points are generally considered to be a good basis for the on going studies but the increasing volume of documentation and related discussion is making it more difficult to locate the information. This Annex contains a first draft which may be used as a common basis for further study of the ISDN and customer access arrangements.

For this Annex the points have been largely extracted from existing Study Group XVIII documentation, modified by discussion where necessary, and reproduced. It is intended that an edited and possibly extended version should be made available for the April 1980 meeting of Study Group XVIII, and later annexed to the ISDN Questions for the next study period.

The points in this Annex are divided into different aspects (network, signalling, transmission, interfaces and terminal) and are mainly related to the emerging 64 kbit/s IDNs.

The material consists of various views and ideas that have been presented in documentation. They do not constitute an agreement that this is the way an ISDN will develop. In fact several of the points may be inconsistent, incomplete or not mutually exclusive.

Further studies are required to consolidate these ideas and to formulate agreed principles upon which further work may be based. It is particularly important that interface and signalling concepts be agreed early so that the detailed development of appropriate Recommendations may be undertaken.

Future studies are expected to provide clarification and elaboration of these points, leading towards the establishment of the main principles for ISDN. It is acknowledged that new techniques and technologies may emerge, together with further network and service option, and these will lead to the establishment of additional points as and when the need arises.

N1 **The telephone network will evolve towards an Integrated Digital Network (IDN) with switched 64 kbit/s telephony channels and it was expected that other services would be integrated with that network during evolution towards an Integrated Services Digital Network (ISDN).**

N2 **The introduction of new services must not prejudice telephony which was expected to predominate indefinitely.**

Dependent on the need of the individual subscriber, flexibility should exist to provide various access types e.g.

a) many subscribers may require only dedicated access to a particular fundamental service on the ISDN (telephony or data);

b) some customers may require alternative access, on a call-by-call basis, to two or more services;

c) some customers may require simultaneous access to two or more services;

d) supplementary services such as listed in Recommendation X.2 and possible in SPC telephone exchanges should be possible in a harmonized form for data and telephony (e.g. network recall, closed user groups, call deviation etc.);

e) new service forms such as with multi-purpose terminals including the capability to change from voice to non-voice or the simultaneous use of both are to be expected.

N3 **Flexibility is needed to permit progressive introduction of various new services, many of which are not yet well defined. The objective would therefore be to establish network capabilities for subsequent exploitation.**

N4 The ISDN access arrangements should be optimized for digital services although due account should be taken of the large and continuing population of analogue telephone and lines

N5 **In studying features of customer access attention should be paid to the usefulness of structuring the interface in functional levels in order to avoid unnecessary constraints on technological evolution. It is** important to establish a level for the common telephony interface, as a basis for the multi-service ISDN terminal.

N6 While it is recognized that customers will have access to a local concentrator or exchange, and that the local exchange will provide many of the basic telephony and perhaps other services, some specialized services need not be provided by all local exchanges. The concept is that local exchanges can provide access, on a call-by-call basis, to other exchanges in the network which can be regarded as special nodes because they provide additional service capabilities. Access to such nodes could for example be on demand following class of service or selection information, or on a hot-line basis for every call from a particular customer. One example of such use of the ISDN would be in the provision of access to a data packet switching exchange. It is recognized that this concept would significantly influence the studies of customer access to the ISDN.

N7 During future studies, some considerations should be given to the possible use of transmission rates lower than 64 kbit/s (e.g. 32 or 16 kbit/s) for telephony paths in the network, and the effect such paths might have on other customer services in the ISDN.

N8 During future studies, some consideration should be given to the possible use of bit saving techniques e.g. differential encoding for providing "wideband speech" at 64 kbit/s.

N9 Future studies of ISDN access should take into account the probability that :-

a) **Integrated Services Digital Networks (ISDN) for telephony, data and other services will be growing rapidly in the next few years;**

b) **the cost of extending digital transmission to the subscriber's location can be expected to become progressively lower;**

c) **new services and facilities can be provided by taking advantage of the inherent characteristics of the ISDN;**

d) **new signalling arrangements will be needed both from exchange to subscriber and subscriber to exchange;**

N10 a) There is a need to develop within a common framework coherent arrangements for customer access to different services provided by the ISDN;

b) such arrangements have to allow various network architectures, taking into account the varying conditions in different countries;

N11 In an Integrated Services Digital Network (ISDN) a connection may be required for a call which:-

i. is exclusively voice (V) service, including unstructured information;

ii. is exclusively Digital Non-Voice (DNV) service;

iii. changes from Voice Service to Digital Non-Voice Service or vice-versa during the call to provide alternative services.

iv. voice and non-voice services on a wholly digital connection to provide services which, to the user, appear to be simultaneous;

N12 "Another aspect needing an early decision is that of continuous transmission of signals on the subscriber line, even under idle conditions between calls. Some signalling methods and supplementary services (e.g. burglar alarm etc.) may only be implemented when the subscriber line transmission system is operating continuously.

N13 Some administrations are preparing plans for extensive introduction of digital transmission and switching equipment, and studying ways of exploiting the emerging digital network by extension of the IDN capabilities down to the premises of customer who can utilise those capabilities. These networks will not be extensive in the early years but suitable standards are needed to encourage steady evolution.

N14 The national conditions for service integration may differ between countries and evolution towards an integrated services network will vary accordingly. In order to provide a common basis for international standardization of digital networks, however, it is essential to adopt a common strategy for developing an ISDN from the telephony IDN.

These varying conditions give rise to the following objectives:

- introduction of new services and facilities should not give significant cost penalty to the major telecommunication service in the network

- as introduction of new services and facilities will be a continuous process, the approach to service integration should be open-ended

N15

__Customer access types__

Customer terminals and digital PABX will evolve rapidly with technologies like microprocessors, automation in the office sector and the definition of new services. An appropriate definition of the customer access to ISDN is therefore of utmost importance. Whilst aiming at a flexible and openended approach to new services and customer terminals, a limited number of customer access types should be standardized. In a first phase of CCITT studies the following access types are proposed; see also Figure 1.

N16 Basic ISDN access type

This comprises :-

a) a 64 kbit/s digital path for :

(i) digital voice at 64 kbit/s , or

(ii) digital data including Recommendation X.1 data user classes with rate adaptation up to 64 kbit/s , or

(iii) combined digital voice and non-voice at 64 kbit/s (e.g. 56 kbit/s digital voice together with 8 kbit/s data)

While some customers will require dedicated access to only one of these service options, provision should be made for alternative operation on a call by call basis, and for changing during an established call. Limitations to this latter feature will be encountered in mixed analogue/digital networks.

b) A separate digital path at a rate much lower than 64 kbit/s for :

(i) customer/network signalling for services in (a); and

(ii) other services such as telemetry information (e.g. customer alarms); and

(iii) spare capacity for future requirements.

An example of the basic ISDN access type incorporating the features listed above is given in Figure 2. It should be noted that some additional bits have been provided for alignment purposes for use when the transmission system does not provide alignment.

N17 Access at rates lower than 64 kbit/s

Consideration should be given to the possibility of customer access at rates lower than 64 kbit/s (e.g. 56, 48 and 32 kbit/s).

N18 ISDN access type with additional channel option

In addition to the basic system capabilities indicated in N16, some consideration should be given to the provision of other services simultaneously with the functions in a) and b) of N16.

Additional digital paths could be provided to give further services as the cus[illegible] res them. The customer network signalling for these services would be carried within channel b) (of n16). The access structure should be capable of providing a smooth growth of customer services from those indicated in a) and b) (of N16) to multi channel services such as those required by, for example, multi function terminals, multi-slot terminals and PABXs.

The additional digital paths may be adapted to 64 kbit/s where necessary (e.g. for Recommendation X.1 services) at the exchange and switched through the digital network.

An example in which the additional access provides a low-rate data service is given in Figure 3 (see also N19).

N19 Treatment of services requiring more than 64 kbit/s

Two approaches have been identified :

(i) services requiring only a few (e.g. < 5) 64 kbit/s digital paths might utilize parallel subscriber access at 64 kbit/s. Further study is needed to determine the network control signalling arrangements for such applications;

(ii) as an alternative to (i) a small transmission multiplex of n x 64 kbit/s could be used;

(iii) for services requiring more (e.g. > 5) 64 kbit/s digital paths, a multiplex based on the primary multiplex systems would appear to be more appropriate;

N20 Treatment of services requiring multiple digital paths (including PABXs)

The approaches identified in N18 also apply in this case.

N21 Treatment of services requiring less than 64 kbit/s digital paths

For services such as low-rate data, the bit stream could be adapted and carried on a standard 64 kbit/s bearer to the exchange. Alternatively a low-rate access comprising a bearer to suit the service with an out-slot channel for signalling etc. (see N16), and with adaptation to 64 kbit/s at the exchange (or intermediate muldex) for switching through the digital network at 64 kbit/s could be used. (Question 29/VII is relevant.)

Another alternative is to interleave low-rate data messages with signalling messages on the out-slot channel of a multi-service access. (See S8). Yet another possibility is to provide access to a packet switching facility and route the low-bit rate data across the ISDN by the use of packets.

N22 Treatment of customers requiring unidirectional transmission

Some provision may need to be made for such services but further study is needed to establish relevant principles. (See paragraph 4 of the Report.)

N23 Functional network architecture

In order to deal with ISDN in an orderly manner it should be given a suitable functional structure. A first approach is to divide the network into two categories of functions:

- a category of basic functions relating to circuit switched 64 kbit/s digital connections which would be provided by all local exchanges; this category provides telephony and possibly circuit switched data service

- a category of additional functions relating to services which require additional capabilities which need only be provided in special equipments located in particular network nodes. The digital local network would, as part of the basic functions, provide access to these special equipments.

N24 ISDN network aspects

It is expected that the ISDN will use 64 kbit/s bearer digital paths. The bit integrity of customer generated information should be preserved between customer terminals. Voice frequency signals should be encoded according to Recommendation G.711.

N25 Network control signalling between the customer terminal and his local exchange for setting up 64 kbit/s connections should be carried outside the 64 kbit/s channel. Note that this does not preclude the use of the 64 kbit/s for some customer/network and customer/customer signalling after the initial out-channel signalling procedure. (See N16)

N26 During the evolutionary stages where extensive interworking occurs between the new digital equipment and existing switching equipment for telephony type traffic; the new network equipment should ensure that ISDN calls are routed appropriately. Where routing over digital facilities only is required, adequate identification and signalling capabilities should be provided in order to remove any service restriction on a particular connection. Note that this requirement should include the identification of apparently digital circuits routed via transmultiplexers or time assignment or other digital processing equipment which restrict the bit sequence integrity of the transmission path.

N27 The ISDN should include adequate provision for services which normally operate at rates other than 64 kbit/s. These include :

- Low rate data;

- Low rate non-standard encoded voice;

- Higher rate services which may be routed through the network on n x 64 kbit/s connections;

- Wideband services requiring transmission paths much in excess of 64 kbit/s (e.g. 2 Mbit/s or more) which, if connected over a network separate from the ISDN, may use the ISDN for associated setting-up and communication purposes.

- Very low rate telemetry over the customers local line as a means of conveying alarm and surveillance communication to a central location such as the local exchange site.

N28 In order to ensure adequate performance and flexibility the ISDN shall be comprised of SPC digital local and higher order exchanges, interconnected by digital transmission and employing fast inter processor common channel signalling between exchanges, CCITT No.7. There will, however, be freedom to move functions between nodes in the ISDN hierarchies, and it is expected that some nodes will be equipped with extra capabilities to satisfy the special requirements of calls routed to such nodes. Quality of service standards for ISDN traffic must be specified in terms of error rates, response times, grades of service etc, and the need for preferential or priority treatment of one service with relation to another on a call by call basis should be considered.

N29 Possible implementations showing different degrees of integration

a. Minimum integration :

In this case the out-slot signalling technique (See S2) (level 2) and the setting-up of digital circuit switched 64 kbit/s connections are primarily designed for telephony needs and a separate in-slot protocol is used for data call set-up. is implies that a hot-line type of connection is used for data calls to establish an access path between the data terminal and the network functions controlling data call set-up consistent with Recommendations X.21, X.25 and the new FDTE interface (also referred to as the multipurpose interface for non-voice services).

Adoption of this approach would imply that a deeper integration of access protocols, and thus call set-up control in the network, can later only be achieved after introduction of new or modified access protocols.

b. Medium integration .

In this case an out-slot signalling protocol (OP) is used for controlling setting up of digital circuit switched 64 kbit/s connections for both telephony and data. This implies that an out-slot signalling technique that is suitable both for telephony and data is chosen. The CTE*) should then provide insertion of the data signalling protocol generated by the DTE*) into the out-slot signalling link of the subscriber line. (*) See Figure 1.)

The out-slot signalling link would carry information blocks which include Service indications to allow separately defined telephony and data versions. These versions should naturally be defined with maximum commonality. A comparison may be made between such versions and the Telephony and Data User Parts in CCITT Signalling System No. 7.

On the assumption that commonality is achieved between telephony and data for the out-slot signalling functions, this degree of integration would provide a good basis for moving to a deeper degree of integration at a later stage. The restrictions for deeper integration in the ISDN would in this case not be imposed by the access protocols but by the fact that call control functions in an IDN evolving to an ISDN has to cater for historical differences in call handling between telephony and data, which in the longer term are likely to be minimized.

The medium integration commonality between telephony and circuit switched data is likely to be achievable with the multi-purpose interface for non-voice services. However, circuit switched data according to Recommendation X.21 and packet switched data according to Recommendation X.25 will have to be treated as for the minimum integration network architecture.

c. Maximum integration :

In this case a common out-slot protocol (OP) would be used for data and telephony. The parts of the OP that relate to common functions, i.e. control of setting-up of a connection and control of supplementary services/user facilities that have a common purpose, would be identical, and ideally the numbering, call routing and other such basic aspects would be common. Naturally, the OP could include particular portions that relate to supplementary services/user facilities that are only applicable for either o. telephony and data. The differences in the actual OP employed and the call handling in the network, would, however, in principle not be differences between telephony and data but differences between calls involving different sets of supplementary service/user facilities. That is, the nature of differences between telephony and data would be no different from the nature of the differences between calls within each service.

N30 It is recognized that the definition of a multi-purpose interface for non-voice services has a bearing on the system concept for the general digital subscriber line signalling interface. Such an interface should, when medium or maximum integration is applied, be specified for both telephony and data applications. In order to avoid proliferation of new customer interfaces, it is necessary to harmonize the studies on a multi-purpose interface for non-voice services with those on local IDNs. On the other hand it is outlined that data requirements should be defined as early as possible in order to be properly taken into account when developing the general digital subscriber line signalling interface.

N31 The definition of standards for customer access to different services provided by ISDNs should be made in such a way as to allow various network architectures, taking into account the varying conditions in different countries, for example the provision of some services through ISDNs by means of interworking with specialized networks. Such dedicated networks may co-exist with the developing ISDN at least during a transition period.

ASPECTS OF INTERFACES BETWEEN THE DIGITAL LOCAL EXCHANGE AND THE LOCAL LINE OR REMOTE UNITS (D, E, F, G AND H IN FIGURE 1)

E1 Study Group XI is specifying the parameters and characteristics of 64 kbit/s digital switches for telephony both at transit and local levels. In the first place CCITT Recommendations for digital local exchanges should cover the connection of analogue subscriber lines and accept already established national standards for analogue subscriber lines. Interfaces with digital (telephone) PABX accepting national PABX signalling systems shall also be covered.

E2 New Recommendations are required for the interfaces concerning digital subscriber lines to include data, telephony, telemetry, etc., signalling and alignment functions.

E3 Moreover it shall be studied what additions to the parameters and characteristics for digital 64 kbit/s switches specified for telephony, if any, are required for the use of these for data switching.

E4 New Recommendations are also required for exchange interfaces with digital PABX, remote digital PABX, remote MULDEX and other multi-slot services. Note that the subscriber line and multi-slot transmission systems may terminate on a MULDEX or a concentrator remote from the serving centre, and common interface standards should apply as far as possible.

E5 At the present time remote subscriber connecting units are considered to be functionally part of the local exchange. The type and information content of the signalling and/or control channel between the remote subscriber connections unit and the exchange terminal is at present considered not to be a matter for international recommendation. The reasons for this are the complexity of such a recommendation with the variety of realizations presently under study.

Most of the implementations considered at present are system-dependent, but a desire has been expressed (in Study Group XI) by some delegates to define a system-independent remotely located switching unit in the future. In this case, CCITT Signalling System No. 7 should form the basis for the signalling and control channels. The study of the system-independent remotely located switching unit was considered to be additional to the study at present undertaken in Study Group XI.

E6 The digital subscriber line interface should be considered from the functional point of view. The functional description can apply with any technical solution for the transmission format or structure on the line.

E7 The digital subscriber line interface might provide a common 64 kbit/s interface for the users information path for all services. It may, however, be more economical to provide separate interfaces for some services. For n x 64 kbit/s services, n x 64 kbit/s user parts might be applied.

E8 A functional interface for signalling, control and alignment purposes should be defined.

E9 The interfaces for digital PABXs are of great importance. It was considered likely that two different interfaces could be provided, one identical or similar to the digital subscriber line interface for smaller PABXs and another more complex one for the larger PABXs. The signalling systems used in the two cases may differ.

E10 Problems arising from the BORSHT

Functions (B = battery; O = over-voltage protection; R = ringing; S = supervisory; H = hybrid; T = test function) are to be studied.

SIGNALLING ASPECTS

Note : Where possible these points have been extracted from available text, with modifications to take account of subsequent discussions. Some new text has been derived from the Questionnaire associated with COM XVIII-No. 131 and from subsequent replies with the issue of this Annex. The Questionnaire is considered to be out of date and will not be studied further in its present form. Some of the points below may be more appropriate under the heading of Network Aspects in later issues.

Customer/network signalling

S1 Classification of access protocols

Access protocols are proposed to be classified in two types, namely :

- an Out-slot Protocol (OP) for out-slot signalling, and
- In-slot Protocols (IP) for in-slot signalling

In addition to these access protocols, user-to-user protocols and user-to-network resources protocols need to be considered. Some of the user-user protocols may standardized consistent with other access protocols.

S2 Out-slot Protocol (OP)

An out-slot signalling protocol for access to network elements controlling set-up and release of all digital 64 kbit/s circuit switched connections. OP would cover signalling for simple calls and further signalling necessary for control of such supplementary services/user facilities, depending upon the degree of integration. See N20. The OP access link (i.e. the out-slot channel with its level 2 technique) would typically terminate at the first subscriber line concentration stage and the OP signalling would, as applicable, be further conveyed in the network over, for example, common channel signalling links.

S3 In-slot Protocols (IP)

A number of service dependent in-slot signalling protocols for access to additional network functions. In some cases IP would correspond to a second call set-up phase (e.g. in some interworking situations).

IP would also cover cases of communication between the user and a central network function, e.g. for facility registration and cancellation at a special service centre. A simple case of an IP would be transmission of digital tones over the 64 kbit/s channel to a calling telephone subscriber. Another case of an IP would be the application of levels 2 and 3 of Recommendation X.25 to access packet-switching facilities.

PABX/network signalling

S4 Where digital and possibly service-integrated PABX are connected to an integrated services digital network the signalling for individual customers access could be used for smaller PABX whilst common channel signalling seems appropriate for larger PABXs. This means that any study of new signalling arrangements for digital PABX should be performed in conjunction with the studies of signalling on digital subscriber lines and the further studies of common channel signalling.

S5 Studies of a signalling system for customer access to the ISDN should be based on the assumption that digital transmission techniques will be extended into the local network down to the customers terminal, and, where appropriate, the characteristics of the access will be the subject of CCITT recommendations.

S6 Studies of the customer/network signalling system protocols should anticipate the requirements of all services likely to be carried by the ISDN. Due attention should be paid to signalling for a basic digital telephone service while preserving maximum potential for more complex signalling for multi-service and multi-channel terminals. If and where appropriate some allowance should be made for the possibility that the new digital-access signalling system might be adapted to be used also to enhance the services provided by conventional analogue access.

S7 The new customer/network signalling system should include the capability of carrying signals, in either direction, between the customer and the local exchange without interruption of the channel to which the signals refer. This capability will permit 'silent' signalling for telephony customers (e.g. private meters).

S8 When considering customers connected to the ISDN it must be recognized that some will require only dedicated access to a particular service, some will require alternative access - possibly on a call by call basis - to two or more services, while others may require access to several independent services at the same time. These options should be taken into account when defining the new customer/network signalling system and it has been suggested that an out-slot signalling link should be provided for each of the channels used for services. However it is accepted that such signalling links may not exist in 'real' channel terms because they are provided as 'virtual' channels using interleaved messages on a common out-slot transmission channel between the customer and the local exchange. Where the 'virtual' channel method is used, the signalling system must contrain suffient addressing etc. to identify the different channels used for services. In addition, where channels are used alternatively for different services, the signalling system must incorporate a means of identifying the service required.

S9 The digital access to ISDN is expected to include provision for very low-rate data and telemetry services, some of which may be switched through the network. The customer/network signalling system could permit such services to be carried over the out-slot transmission path using either a cyclic multiplex or interleaved messages.

S10 The signalling system should be suitable for use in local networks in which customer accesses are connected to an intermediate digital multiplex to provide transmission to the exchange over standard primary local digital line systems. In these circumstances it has been suggested that the customer/network signalling links, in 'real' or 'virtual' channels will be interleaved onto a common bearer such as T/S16 of the 2048 kbit/s line systems.

S11 The signalling system should be suitable for use with simple terminals involving human operators such as for the telephone service, as well as complex terminals having machine generation and detection of in-slot and out-slot signals.

S12 Consideration should be given to the possibility that the ISDN may be used for services which require unidirectional transmission, such as broadcast, and the effect such services might have on the signalling system and protocols operational checks.

S13 Since it is envisaged that ISDN might carry wider-band services using the multi-slot connection techniques, consideration should be given to the signalling protocols appropriate to such connections.

S14 During the definition of a signalling system to satisfy the requirements for ISDN access, some consideration should be given to the line-feed and other BORSHT functions to establish the extent to which the signalling system may be affected by different implementations of these functions.

TRANSMISSION ASPECTS

T1 Preparation of guidelines on transmission requirements for customer access in relation to a range of customer access types

- Three different possibilities could be identified :

  a. A "single subscriber, single timeslot" requirement of $(64 + \Delta)$ kbit/s in each transmission direction. (Examples can be found in Documents Nos. COM XVIII-No. 310, Section 1.3.1; COM XVIII-No. 265);

  b. A "single subscriber, multi-timeslot" requirement of $n \times 64$ kbit/s + Δ^1 kbit/s in each transmission direction. (An example can be found in Document No. COM XVIII-No. 266.)

  c. A PABX connection, requiring $m \times 64$ kbit/s + Δ'' kbit/s in each transmission direction (COM XVIII-No. 243).

T2 Study of bit stream structures to allocate transmission capacity and provide appropriate alignment

- Different structures will be required for the classes under Sections 3.a and 3.c, while the class mentioned under Section 3.b can use either a parallel combination of "3.a-systems" or a "3.c-system".

- Delayed Contribution BT indicates some general outlines for possible structures for class 3.a.

- The subscribers loop transmission system has to provide for the transmission of sufficient information between the exchange and the subscribers terminal in order to make demultiplexing of the various channels possible. (An example has been indicated in Section 13 of COM XVIII-No. 310.)

T3 Transmission systems for access types identified for initial study

- Delayed Contribution BT (Section 2) indicates that the required functions can be provided with either 2-wire or 4-wire transmission systems.

- The Report of Working Party XVIII/4 (Contribution COM XVIII-No. 312) indicates the power feeding of the subscribers terminal as an important study point.

T4 Transmission system for connecting PABXs

- The required transmission systems probably are almost identical to the already specified 1544 kbit/s /2048 Kbit/s systems or small capacity digital transmission systems, e.g. 11×64 kbit/s (704 kbit/s).

  No information, however, on this subject could be identified in the available documents. The need for further study has been indicated in the report of Working Party XVIII/4 (Contribution COM XVIII-No. 312).

T5 Transmission standards applicable to remote access to the ISDN

- No relevant information could be found in the available documents.

T6 During discussions in Working Party XVIII/1 it became clear that no Administration yet wished to specify the subscriber line transmission system itself.

CUSTOMER INTERFACE ASPECTS

C1 Working Party XVIII/1 has so far identified interface types as shown in Figure 1. Not all of these interfaces will necessarily be the subject of CCITT Recommendations. Some are already specified in CCITT Recommendations and these should be taken into account. The aim should be to define a very small number of these interface types, which may not necessarily appear as physical interfaces.

C2 It is desirable to define as small a number of interfaces of types A, B and C as possible, while allowing for all types of STE provided to meet customer requirements.

C3 The digital local network may, seen from the customer as part of his information system, carry the following information transactions and provide the following services :

- speech connections
- supplementary telephone services
- data connections to subscribers in separate data networks or data equipment in the digital network; for connections to separate data networks the connection must be compatible with existing standards for circuit - or packed-switched services and existing user classes
- a text and control connection to local or remote information data-bases, teletax, videotext system etc (possibly in parallel to the speech connection)
- end-to-end signalling and control connection with the remote subscriber for session control (terminal configuration control, protocol selection, input/out selection)
- receiving/sending of text and other non-voice messages to subscriber terminals, including the case of calls arriving at a terminal which is unattended or engaged on another call
- alarm service capable of carrying alarms from sensors etc. independent of communication on the subscriber line
- advanced charging information capability during call.

C4 It is expected that the data requirements will greatly influence the customer interface and work on the standards should take account of those applicable for circuit and packet switched data transmission (Recommendations X.21 and X.25). The interface specification should be structured, and based upon the levels being identified for data. For example, the interfaces of the digital subscriber line could be organized with the following functional levels :

Level 1 - Physical link provision defining the functions relating to the physical, electrical, logical and line transmission technique related parameters of the interface (as applicable).

Level 2 - Link access protocols defining the functions relating to, for each communication channel provided by level 1, the technique by which higher level information is transferred over the channel.

Level 3 - Call control procedure defining the functions relating to control of call set-up and release for switched services.

Level 4 - End-to-end protocol defining the functions relating to end-to-end communication between subscriber terminals or between subscriber terminals and network resources.

Depending on the type of service and how it is accessed the level 2, 3 and 4 elements may have to be defined separately for different phases of a call.

SERVICE ASPECTS

M1 In considering interface A at the subscribers premises to be the user's access point to the ISDN, it is recognized that the link to the exchange may be shared for several services. There are many network structures possible to provide multiservice compatability. Figures 4 and 5 show various examples.

a) The telephone network and the dedicated networks are separated at the distributing frame.

Each service having its own signalling system.

b) The telephony network and the dedicated networks are separated at the local switching exchange. The multiservice signalling system is used.

- For the telephone signals

- For the seizing signal, the clearing signal and some test signals for the other services.

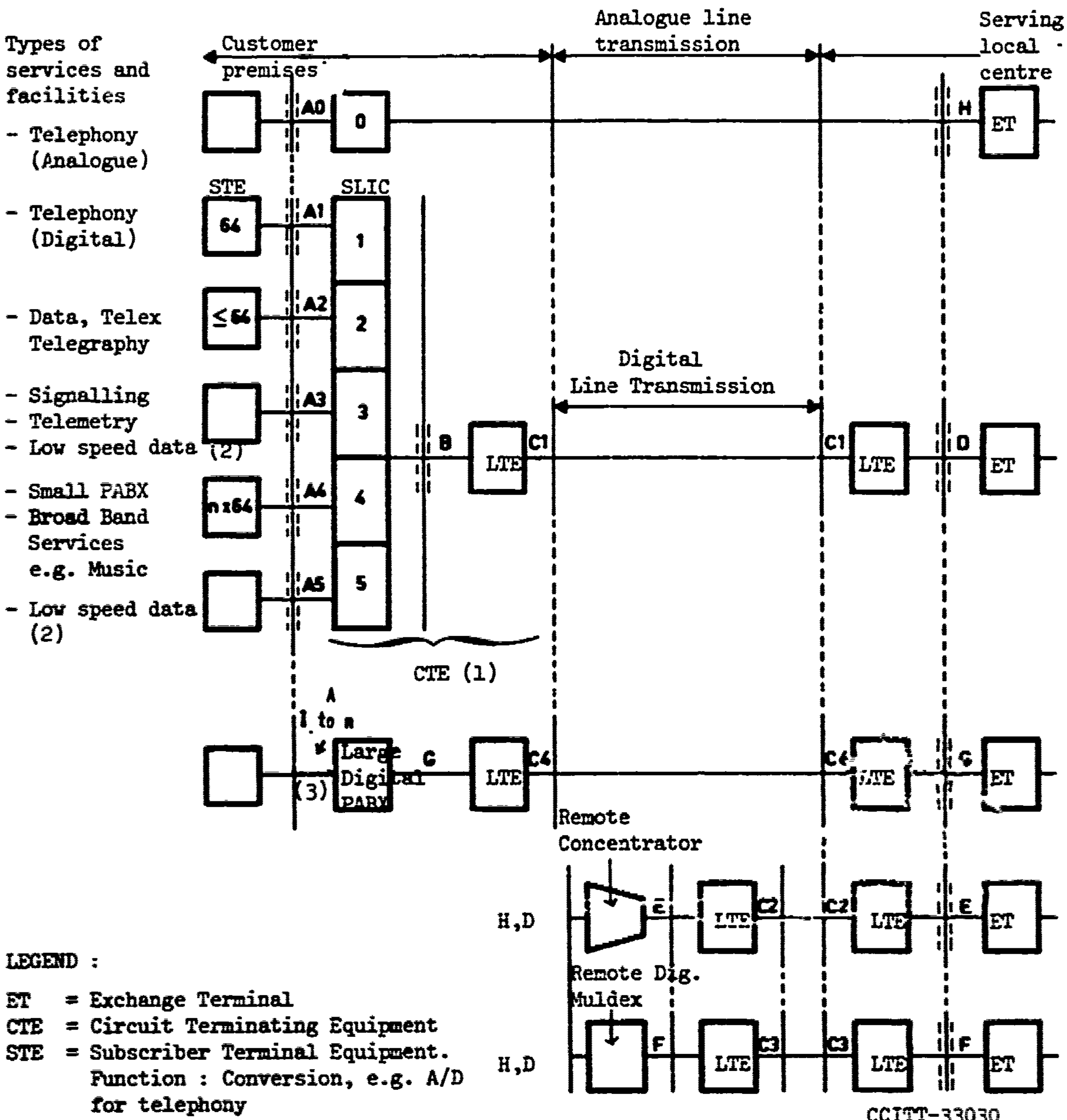
c) The multiservice Network is completely integrated but this does not exclude some dedicated service Nodes (processing of Data Transmission). A single multiservice signalling system is used for all services.

The configuration described above may follow each other in a transition period.

M2 While the interworking of the ISDN evolving from the telephone network with co-existing dedicated networks will lead to the offering of the wide range of existing services (e.g. telephony, X.1 and X.2 data services, packet switching) on the new link from the subscriber to the exchange, also new services may emerge as new features of the homogenous digital network.

Higher rate capability may be made available by using n x 64 kbit/s connections through the network. Conference calls connecting several subscribers may be required.

The division of the functions necessary for the services between the terminal on one hand and the network on the other hand is likely to change since new technology (micro-processors) leaves scope for more sophistication in terminals and in the network.



LEGEND :

ET = Exchange Terminal
CTE = Circuit Terminating Equipment
STE = Subscriber Terminal Equipment.
Function : Conversion, e.g. A/D for telephony
SLIC = Subscriber's Line Interface Circuit.
Function : Signal arrangement for Line Transmission
LTE = Line Terminal Equipment. Function : Conversion to Line-code and Line-feeding

⁞⁞ = External or Internal Interface

(1) = SLIC and LTE functions may be combined in a CTE
(2) = Optional data access
(3) = The value of n may be greater than 5 to permit PABX options not necessarily available on A1 to A5

A, B, C, D, E, F, G, H = Interfaces, some of which may need to be the subject of CCITT Recommendations

Figure 1 - Proposed interfaces in local networks for ISDN

CUSTOMER ACCESS FUNCTIONS				EXCHANGE TERMINAL FUNCTIONS
	CHANNEL	BIT RATE	ALIGNMENT	
(a)	Voice at 64 kbit/s or Data at ≤ 64 kbit/s or Combined voice/data (e.g. at 56+8 kbit/s)	64 kbit/s	≤ 8 kbit/s	Switching at 64 kbit/s
(b)	Customer/network signalling (2 kbit/s) and Telemetry (≤ 1 kbit/s) and Spare capacity (≥ 5 kbit/s)	≤ 8 kbit/s		Signalling termination Telemetry termination Further services termination (?)
CUMULTATIVE TRANSMISSION REQUIREMENTS		≅ 72 kbit/s	≤ 80 kbit/s	

Figure 2 - Example of a basic ISDN access type

CUSTOMER ACCESS FUNCTIONS				EXCHANGE TERMINAL FUNCTIONS
CHANNELS	BIT-RATE	CUSTOMER-NETWORK SIGNALLING	ALIGNMENT (IF REQUIRED)	
Voice at 64 kbit/s or Data at ≤64 kbit/s or Combined voice/data (e.g. at 56+8 kbit/s)	64 kbit/s			Switching at 64 kbit/s
		2 kbit/s		Signalling termination
Data at 8 kbit/s	8 kbit/s			Adaptation followed by switching at 64 kbit/s
		2 kbit/s		Signalling termination
Telemetry	2 kbit/s	nil		Termination with other telemetry channels on local muldex
			2 kbit/s	
CUMULTATIVE TRANSMISSION REQUIREMENTS	74 kbit/s	78 kbit/s	80 kbit/s	

Figure 3 - An example of ISDN access with an additional data service channel

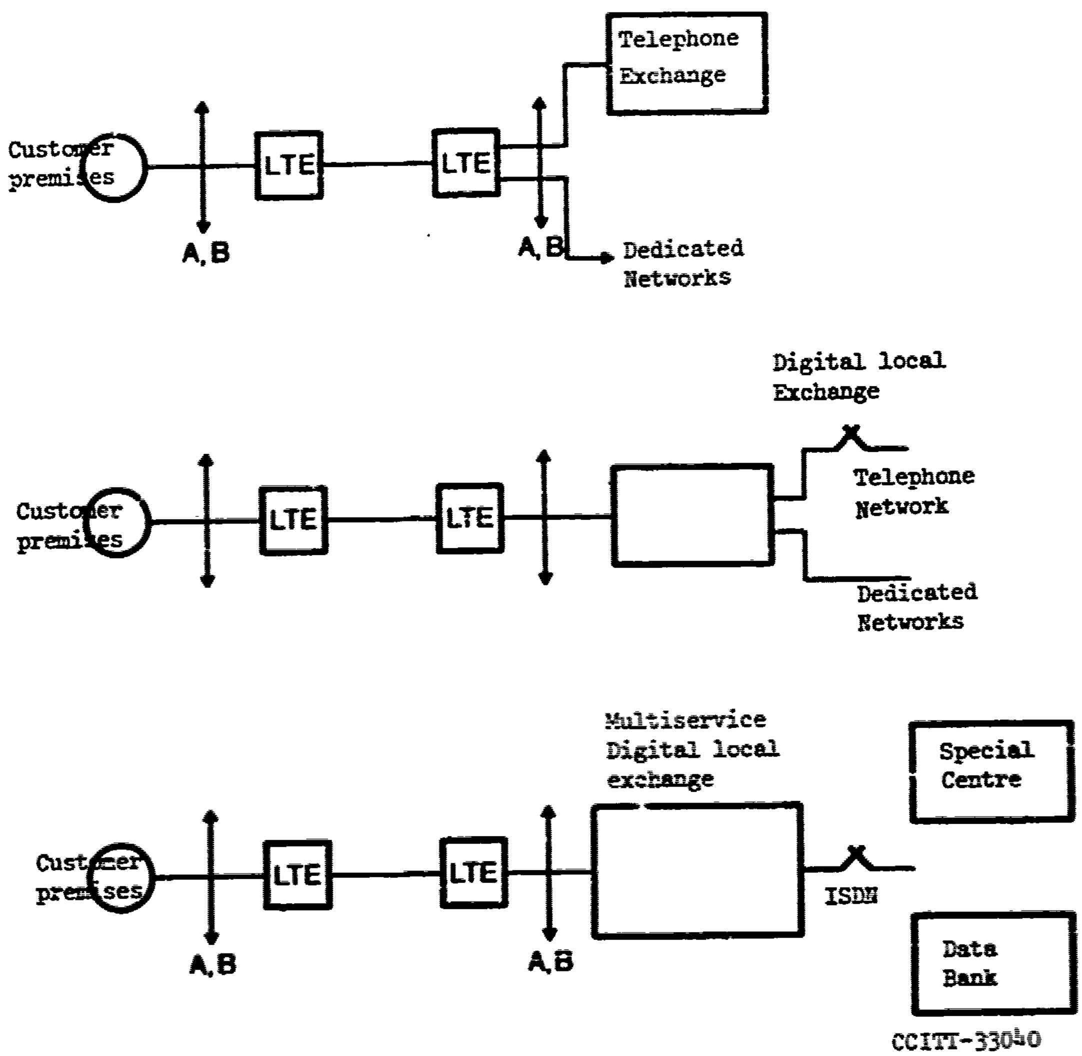
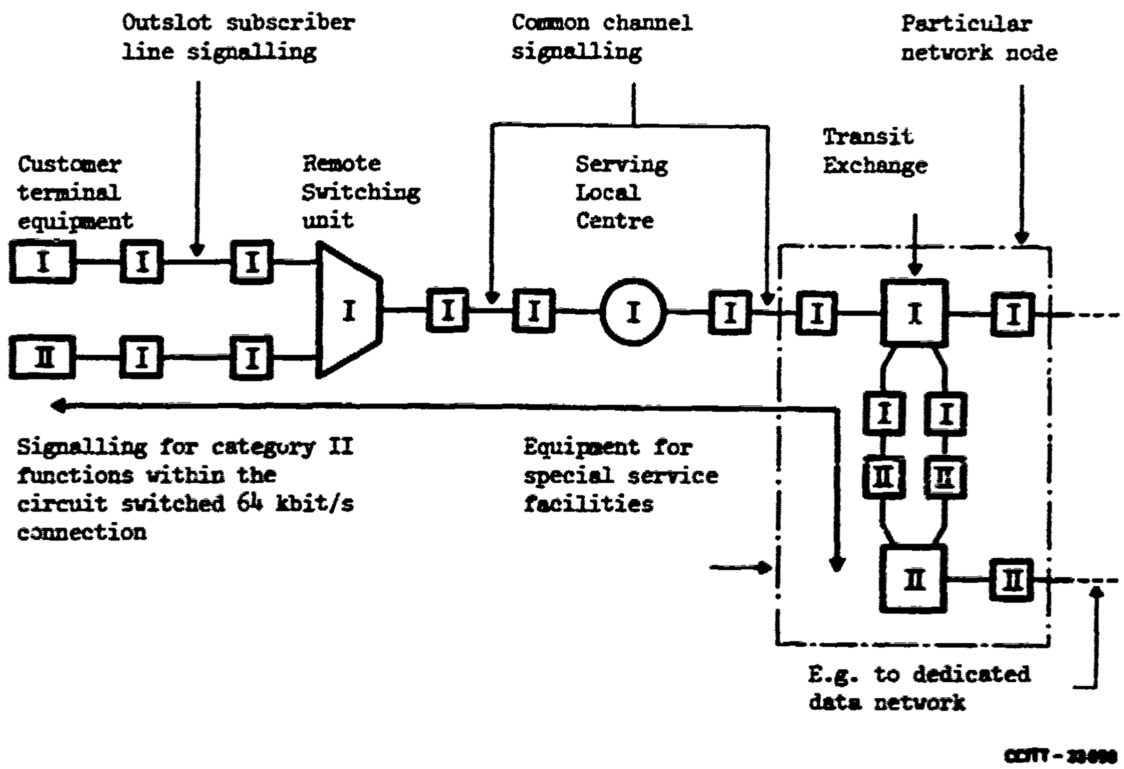


Figure 4 - Examples of multiservices network structure



Category **I** : Basic functions for circuit switched 64 kbit/s digital connections, typically for telephony and possibly for circuit siwtched data. These control and signalling functions are provided in all terminals and exchanges.

Category **II** : Additional special service control and signalling functions are provided in special terminals and equipments.

Figure 5 - Concept of two categories of network functions

Annex 3

(to Question 1/XVIII)

Digital subscriber line signalling

(Contribution from American Telephone and Telegraph Company)

1. Scope of the Contribution

An overview of possible information and signalling structures for a Digital Subscriber Line is presented leading to a channel-level characterization of a generalized Digital Subscriber Line. This specific Digital Subscriber Line structure is not proposed, but should be an element of the set described by the characterization. The requirements upon a signalling means to support the generalized Digital Subscriber Line are summarized and several specific proposals for Digital Subscriber Line signalling are presented. These include the signalling channel bit rate, the use of a bit oriented protocol at the link level, and the use of message oriented signalling. A framework for a layered protocol for Digital Subscriber Line signalling is proposed, with recommendations for further study and contribution.

2. Introduction

An evolutionary process is underway toward an all-digital telephone network, including digital connectivity into the subscriber's premises. This process began with the introduction of digital inter-exchange transmission and is now being extended into both the transit and local exchanges. With the addition of digital subscriber lines this all digital network can become the vehicle for many new customer services based upon a direct digital access capability. This ultimate all digital environment will become the Integrated Services Digital Network (ISDN).

The general attributes of the network will be

1. A functional channel structure whereby identifiable channels can be realized within a uniform digital format.
2. The availability of 64 kbit/s channels between the subscriber and the network which may carry both digitized voice and nonvoice communication.
3. Common channel signalling to implement all inter-exchange signalling, separate from the channels available for intersubscriber communication.
4. The integration of non-voice transmission capabilities and services derived therein with facilities used for voice communication.

A Digital Subscriber Line connecting the subscriber digital station equipment with the network must provide a flexible and robust access to the network. This will be realized through the extension of the 64 kbit/s channel connectivity to the subscriber, as well as the definition of a new signalling interface. This contribution will describe in general terms a Digital Subscriber Line under study by AT&T and will focus specifically on the signalling portion of the Digital Subscriber Line.

3. General requirements for a subscriber digital line

Because the opportunities afforded by the digital connectivity include many features that are new to telephony it is best to state the overall functional needs to be met by the Digital Subscriber Line before proceeding to details of an implementation. Therefore, the Digital Subscriber Line must provide:

1. Basic voice-band telephone capabilities. The familiar call set-up/disconnect signalling functions must be provided with a subscriber-to-subscriber voice path being established. The voice path shall be 64 kbit/s compatible with Recommendation G.711. The Digital Subscriber Line itself shall provide a 64 kbit/s voice channel between the subscriber station equipments and the digital local exchange for this purpose.

2. Circuit-switched 64 kbit/s digital connectivity for non-voice communication. This can be provided via the same Digital Subscriber Line channel used for voice communication on an alternate use basis or through additional 64 kbit/s channel(s).

3. Signalling means to establish either voice or non-voice calls on the 64 kbit/s channel(s). The signalling function proposed herein is implemented via a channel separate from those previously defined. The use of a separate channel enables the definition of new services to the subscriber which require signalling simultaneously with voice or non-voice communication as required to implement multiple channel call set-up, or polling based features such as security services, etc. Only one signalling channel will be specified per Digital Subscriber Line regardless of the number of additional voice or non-voice channels.

4. Provision for low traffic density , long holding time, communication associated with new non-voice services. Many new service capabilities, such as electronic security, electronic funds transactions, enhanced directory services, and limited data base access have the characteristic of interaction between the subscriber at his terminal equipment and a network (or non-network) resource. These interactions may be most efficiently provided by the network through a message switching capability rather than through a circuit switched path.

5. For a Digital Subscriber Line implemented using very wide bandwidth transmission media (such as fiber optic links) the possibility of the inclusion of video bandwidth channel(s) should be accommodated. In particular, the signalling plan should be extensible to control such channels.

A general definition which might cover the specific realization of a Digital Subscriber Line which meets these requirements is characterized by

$$\text{Digital Subscriber Line} = nV + mD + pW + S + F,$$

where V represents 64 kbit/s voice or non-voice channels. The specific Digital Subscriber Line will include n such channels where $n \geq 1$.

D represents < 64 kbit/s channels to be used exclusively for non-voice communication. The Digital Subscriber Line may contain m such channels where $m \geq 0$.

W represents high-rate channels. A fiber optic Digital Subscriber Line might contain p such channels where $p > 0$. The specific transmission mechanism associated with high-rate or video channels, whether analogue or digital, cannot be established at this time.

S represents a message oriented (as distinguished from circuit switched) channel upon which all signalling to and from the subscriber's premises takes place.

and

F represents framing, synchronization and other line control bits. These signals are not available to either the subscriber or exchange for information transfer.

A functional diagram of the Digital Subscriber Line, the Digital Local Exchange and the subscriber premise equipment is as depicted in Figure 1.

The Signalling Channel provides for the coordination and control between the exchange and the subscriber. In general, the subscriber equipment might consist of a centralized subscriber control equipment and a multiplicity of connecting subscriber terminal equipments. In order to facilitate this control function, the signalling channel must have the capability to identify call activity on the remaining Digital Subscriber Line channels, and to identify the specific subscriber terminal equipments involved in such activity. As such, the functional needs for signalling are very similar to those well known for communication between a pair of local exchanges. A channel numbering plan and a terminal numbering plan must be included.

4. Aspects of a digital subscriber line subject to Recommendation

It should be the objective of Recommendation to encourage flexibility where growth and change is desirable, and to fix certain parameters where growth or change would render major portions of a digital network obsolete. The possibility of a family of specific Digital Subscriber Lines with various number of V, D, and W channels should be kept open because future service opportunities may require additional bandwidth. Future technology, especially fiber optics, may provide the high bandwidth Digital Subscriber Line connectivity in a cost effective manner. However, the specific communication protocols associated with Digital Subscriber Line signalling should be established and held reasonably constant in order to minimize the impact of evolutionary change upon the network signalling facility. It is therefore desirable to establish a highly flexible Digital Subscriber Line signalling plan which is both general and extensible and independent of the transmission method and medium to provide a built-in mechanism for growth and change. Such a goal may be realized through the use of a message oriented signalling protocol following generally in method and philosophy from that established for inter-exchange common channel signalling.

The Digital Subscriber Line, therefore, should be defined given a minimum configuration with opportunities for further extension in the areas of voice and/or non-voice channels. The minimum Digital Subscriber Line consists, therefore, of a signalling channel and a single voice channel. Obvious extensions include additional 64 kbit/s voice/non-voice channels, additional non-voice channels of (for the present) unspecified bit rates, or video rate channels.

With reference to Figure 1, the channels previously described should be defined specifically at interfaces B and E only. This will allow for evolutionary improvements of the transmission and system technologies related to interfaces C and D. Interface A should also become subject to standardization but refers to the intrapremises link between a subscriber terminal and the centralized communications equipment. As such, it is not a component of the Digital Subscriber Line; i.e., it does not interface directly to the network and should be considered separately.

With these considerations in mind, the remainder of this contribution will be addressed to the information structure of the Digital Subscriber Line as specified at interface E. Specific bit rates will refer to the information bit rate of that interface and not to the transmission rate as specified at interfaces C or D.

5. Bit rates associated with the digital subscriber line

The Digital Subscriber Line shall carry signalling, voice, and non-voice information organized into channels and conforming to the expression

Digital Subscriber Line = nV + mD + S.

A bit rate associated with video transmission (W) cannot be established at this time, and will be omitted from further discussion. Also, the specific bit rate associated with < . kbit/s non-voice channels (D) cannot be identified at this time. At interface E (Figure 1) voice or non-voice channels (V) shall be realized at 64 kbit/s. The signalling channel shall be realized at $\geq$ 8 kbit/s.

The particular electrical format defined at interface E is not proposed herein. Only the information to be provided at interface E is specified.

In addition to the above channels, non-information bits will be provided for synchronization and alignment. All such bits used for line synchronization, framing and line integrity appearing at interfaces C and D shall be utilized by the Line Terminating Equipment (LTE) and not reported to interface B.

The selection of a bit rate for the signalling channel must take into account both the average message information bit rate to meet the needs of telephony and new services, and minimum delay requirements with respect to system responsiveness to subscriber input stimuli. Studies have shown that the average information rate (channel occupancy) on the signalling channel will be very low . However, the delays associated with a message oriented signalling protocol would be excessive at low signalling channel bit rates. Therefore, in order to minimize signalling delays in order to meet anticipated STE/ET delay requirements a signalling channel bit rate of at least 8 kbit/s is proposed.

6. Communication protocol associated with digital subscriber line signalling

In order to maximize the flexibility and extensibility of Digital Subscriber Line signalling, the choice of a hierarchical (or layered) signalling protocol is recommended. Schemes similar to Signalling System No. 7 have been considered as models for such a protocol. Because the Digital Subscriber Line interfaces with the customer's premises, the cost of signalling implementation must be kept as low as possible. For this reason, the direct utilization of Signalling System No. 7 for Digital Subscriber Lines does not seem reasonable. However, a new hierarchical protocol tuned to the needs of Digital Subscriber Line signalling is generally described as follows:

Level 1 - Physical Link: A full duplex link of at least 8 kbit/s is specifically proposed.

Level 2 - Link Level Control: Link control procedures using a bit oriented protocol with a frame level structure comprising frame delimiters (flags), zero insertion/deletion, and cyclic redundancy check. This will provide secure message alignment protected from information pattern simulation, complete transparency, and effective detection of transmission errors. Specific error recovery procedures should be tailored to Digital Subscriber Line signalling characteristics.

Level 3 - Routing: Message level control will be required to manage end-to-end message communication.

Level 4 - End-to-End Message Transfer: Messages will be passed from the subscriber control equipment to the digital local exchange requiring little message-transfer control. However, those messages destined for other entities in (or external to) the telephone network will require message transfer control. Further contributions are required in this area.

Level 5 - Application: The end applications consist of the software processes located in the subscriber control equipment and signalling destination which actively initiate and/or receive signalling messages.

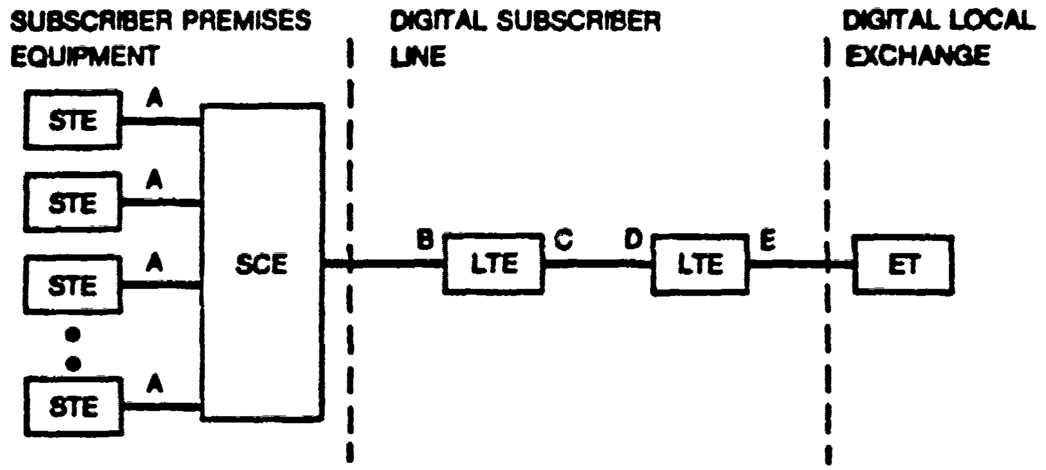
Formal specification of layering beyond level 5 is not proposed in this contribution and may not be necessary because the software in the Subscriber Control Equipment will not be excessively complex. This does not preclude further layering of the application layer at the discretion of the user.

Specific definition of Levels 1, 2, and 3 should be the subject for Recommendations, while the full definition of Levels 4 and 5 should be left to the particular Administration. A basic message set should be defined at levels 4 and 5 to accommodate basic telephony services. However, the full message set will reflect the particular set of services and features to be offered by the individual Administrations and may vary from Administration to Administration.

Message types to be included in Level 5 include signal and maintenance information. Messages should be addressable from the subscriber control equipment to the Local Exchange for normal signalling, for end-to-end terminal control and for special call features.

7. Conclusion

A signalling plan for Digital Subscriber Lines must not be limited by present technological realizations. Future capabilities will surely add continual growth to the requirements of Digital Subscriber Line signalling. An overview of a flexible and extensible signalling protocol under study by AT&T has been described herein for consideration in this area. A major problem area still to be addressed is the interworking of Digital Subscriber Lines with analogue subscriber lines via either digital or analogue interexchange circuits for the end-to-end transmission of push-button telephone generated information.



LEGEND

STE- SUBSCRIBER TERMINAL EQUIPMENT
SCE- SUBSCRIBER CONTROL EQUIPMENT
LTE- LINE TERMINATING EQUIPMENT
ET- EXCHANGE TERMINAL

Figure 1 Functional Interfaces for Digital Subscriber Line

Annex 4

(to Question 1/XVIII)

Studies of digital local networks

(Contribution from Swedish Administration)

Introduction

In contribution COM XI - No.277/COM XVIII - No.310 the Swedish Administration presented the results of discussions among some European countries on a possible basis for international studies of digital local networks. The same contribution was available as delayed document D.14 at the April 1979 meeting of Study Group VII. Extracts of the document are included as the Annex to draft Question 1F/VII (see COM.VII - No.369, p.240).

These discussions have continued and resulted in an updated version of the report already submitted. Especially, sections 1.3 and 1.4 of the first report have been replaced by new sections 1.3, 1.4 and 1.5. For updating of the information submitted in the first report these new sections 1.3, 1.4 and 1.5 are reproduced in this contribution. The report is a result of an expert group in which the following Administrations participated:

Belgium, Denmark, France, Finland, Germany (Federal Republic), Italy, Netherlands, Norway, Spain, Sweden, Switzerland.

1.3 Functional network architecture

1.3.1 Concept of two categories of network functions

A first approach is to identify two categories of network functions:

- a category of basic functions relating to circuit switched 64 kbit/s digital connections which would be provided by all local exchanges; this category provides telephony and possibly circuit switched data service

- a category of additional functions relating to services which require additional capabilities which need only be provided in special equipments located in particular network nodes or provided by specialized sub-networks. The digital local network would, as part of the basic functions, provide access to these special equipment or sub-networks. Such access would be on demand and on a call-by-call basis following class of service or selection information or on a hot-line basis for every call from a particular customer. Signalling between the customer terminal and the special equipments would be inslot and independent of the basic functions.

Figure 2 illustrates this concept of two categories of network functions. Further activities are required to achieve a more precise and detailed illustration of different aspects of the (functional) network architecture. Also, a suitable terminology for describing different phases in network evolution needs to be established. It is for example desireable that the CCITT definition of ISDN is modified to take evolutionary aspects into account.

1.3.2 Interfaces, levels

With regard to the different interfaces of the digital local network (see Figure 1) it will be required to define for each interface a number of interface elements relating to the technique adopted and the customer access protocols for different services. These interface elements should be organised in functional levels corresponding to for example the open systems architecture being developed by CCITT and ISO. For example, the interfaces of the digital subscriber line could be organized with the following functional levels:

Level 1 - Physical link provision defining the functions relating to the physical, electrical, logical and line transmission technique related parameters of the interface (as applicable)

Level 2 - Link access protocol defining the functions relating to, for each communication channel provided by level 1, the technique by which higher level information is transferred over the channel.

Level 3 - Call control procedure defining the functions relating to control of call set-up and release for switched services.

Level 4 - End-to-end protocol defining the functions relating to end-to-end communications between subscriber terminals or between subscriber terminals and network resources.

Depending on the type of service and how it is accessed the level 2, 3 and 4 elements may have to be defined separately for different phases of a call.

1.4 Customer access types

Customer terminals and digital PABX will evolve rapidly with technologies like microprocessors, automation in the office sector and the definition of new services. An appropriate definition of the customer access is therefore of utmost importance. Whilst aiming at a flexible and openended approach to new services and customer terminals, a limited number of customer access types should be standardised.

1.4.1 Fundamental assumptions

The customer access types defined in this and subsequent paragraphs relate to the functional elements of the local network as shown in Figure 1. A number of basic information types generated and received by ISDN customers have been identified. These types, which are characterised taking type of service, information rate and handling in the network into account, comprise:

Type a - Signals corresponding to a conventional (analogue) telephone subscriber station. i.e. including decadic pulsing or multi frequency signalling (MFPB)

Note No further consideration is given to this type of information in the context of access definition.

Type b - Digital information at 64 kbit/s representing, for example:

(i) digital voice at 64 kbit/s

(ii) digital signals at 64 kbit/s (e.g. Rec. X.1 data user classes with rate adaptation at the Circuit Terminating Equipemnt (CTE) up to 64 kbit/s)

(iii) Combined digital signal at an overall bit rate of 64 kbit/s (e.g. digital voice together with data).

While some customers will require dedicated access to only one of these service options, provision should be made for alternative operation on a call by call basis, and for changing during an established call. Limitations to this latter feature will be encountered in mixed analogue/digital networks (see para 2.1.c).

Type bb - Digital information at n x 64 kbit/s representing, for example,

(i) digitally encoded broad-band audio signals

(ii) high speed non-voice services (e.g. fast facsimile or still picture transmission).

(iii) combinations of (i) and (ii).

Type t - Telemetry informations at very low rate conveying for example:

(i) customer alarms

(ii) signals for remote control of equipment on customer's premises

(iii) remote meter reading.

Type d - Digital information representing low speed services (e.g. according to Rec. X.1 at a maximum user rate of either 4.8 or 9.6 kbit/s; see also 1.4.2). This information may be transferred simultaneosly with the information of type b, e.g.

- for access to a data base

- for transmission of a facsimile picture.

Furthermore, d might represent a spare capacity for uses yet to be defined.

Type s - Basic subscriber-network signalling information channel allowing for the control and monitoring of network resources by the customer (in a broad sense, e.g. including call progress signals, charging indication, maintenance information, etc.).

It is understood that a single, unified signalling system should be defined having the capability to control the access of one b-type information and also of one ensemble like b + d and also of multiple b-type information accesses.

1.4.2 Basic access

The basic digital customer access to the ISDN is defined in relation to Interface B of Fig. 1. It provides for the transfer of information on a channel with the following composite structure.

$$\boxed{b + \Delta}$$

The following aggregate bit rates are foreseen:

- channel b; 64 kbit/s

- channel Δ; 16 kbit/s and 8 kbit/s.

Note - The choice between 16 kbit/s and 8 kbit/s should be further assesed taking into account further considerations on the information type; namely the desirability of providing for such information and the limitation of the rate for such information to 4.8 or 9.6 kbit/s.

Channel b refers to a channel with characteristics defined in 1.4.1 while for channel Δ, two alternatives are set for further study;

(A) represents a common pool dynamically shared, as applicable, by different information types s, t and d.

(B) is divided in a fixed way, by appropriate time division multiplexing techniques, into two subchannels Δ_1 and Δ_2.

Δ_1 is dynamically shared by information types s and t

Δ_2 is allocated to information type d.

It is understood that the two alternatives (A) and (B) represent in fact two classes of possible solutions. It is urgently required to evaluate the elements of both classes in order to be able to quickly decide on only one (class of) solution(s).

Note 1 - The requirements for an additional communication channel such as for the d type of information, (A) and (B) alternatives, may also be covered by a second b channel.

Note 2 - With the exception of one administration, the view was generally expressed that information on the Δ_2 channel will be padded to 64 kbit/s prior to switching in the exchange.

The agreed common features of the Δ channel in version (A) and the Δ_1 channel in version (B) are:

- Signalling in the form of messages of variable length
- Each message will be framed
- Frames will be delimited by appropriate flags
- Signalling protocols should be bit rate independent
- Signalling protocols should be structured into a number of functional levels or layers (see 1.3.2).

In alternative (A) where d type of information when used for some packet switched data service, will be message interleaved with signalling information, the intention is to specify a comparatively simple link access protocol (level 2) primarily designed for signalling requirements; thus the full capabilities (and complexities)of forexample,the X.25 interface, level 2, will not be provided in the signalling access protocol.

In the following the term "out-slot channel" will be used to denote the Δ or (Δ_1 + Δ_2) channel.

1.4.3 Broadband access

Three cases of access for bb-type information have been identified:

(i) In case the ISDN allows for multislot switching of n x 64 kbit/s (n being a small integer), appropriate C and D type interfaces must be defined, preferably in line with other standard multiplex interfaces.

Note: The multislot switching capability is for further study.

(ii) If (i) does not apply, a few parallel basic accesses according to 1.4.2 may be provided. Further study is needed to incorporate mechanisms within the CTE which ensure time slot sequence integrity.

For both (i) and (ii), the agreed subscriber network signalling method id identical to the one described in 1.4.2. Further study is needed for the practical implementation.

(iii) For services requiring several 64 kbit/s digital paths an appropriate standard multiplex for subscriber access will be provided. The signalling arrangement for these applications are for further study.

1.4.4 PABX access

Two cases have been identified:

Type 1 - Access of the PABX to the serving local exchange is either by by a number of individual basic accesses or alternatively through a suitable multiplex.

Type 2 - Connection of the PABX to the serving local exchange is performed by an appropriate standard multiplex, with common channel signalling. The Message Transfer Part of the signalling shall be consistent with the one defined in CCITT Signalling System No. 7.

1.5 Explanations to Figure 1

Figure 1 shows a number of possible functional interfaces in the local networks of ISDN, some of which may need to be the subject of recommendations. It is emphasised that the block diagram represents functional entities and is not intended to represent hardware implementations.

The following comments apply to Figure 1.

a) The functional circles diagram within the large SLIC box represents the routing of information types into the 64 kbit/s basic channel and the out-slot channel.

b) The SLIC functional representation in Figure 1 does not imply that all functions may indeed always be implemented, e.g the out-slot channel access for low speed services.

c) A certain degree of flexibility is recommended when interpreting the various functional interfaces of the SLIC: A_1, A_2, A_3 and B_0. While interfaces A_2 and A_3 will typically correspond to external equipment interfaces, interfaces A_1 and B might correspond to internal interfaces only. Interface A_1 typically refers to the physical connection of a digital vioce channel at 64 kbit/s with digital out-slot signalling. Should this interface correspond to the physical connection of an analogue voice channel then coice coding and digital out-slot signalling generation would be performed by the SLIC. Interface A_2 typically refers to the physical connection of a data terminal equipment (DTE). For example this may in the near future be an X.21 or X.25 type of interface; in this case data padding up to 64 kbit/s and out-slot signalling generation are performed by the SLIC. However, for future data communication this interface might correspond to the connection of a DTE generating data at 64 kbit/s with digital out-slot signalling. Similarly interfaces A_3 might correspond to the physical connection of slow speed DTEs operating with a CCITT standard interface or a new interface.

d) In view of the several types of interfaces A_1, A_2 and A_3 it is recognised that priority should be given in the first instance to the study of interface B_0 where the characteristics of the 64 kbit/s and the channel can be specified regardless of the physical implementation of functions performed by STEs and SLICs.

The SLIC maps the characteristics of the different A_1, A_2 and A_3 interfaces into a common B_0 interface. Interface B_0 requires the specification of various interface levels, including the interleaving technique of the various types of information. It is recognised that the interleaving technique might imply or not imply the provision of alignment information; this information might be carried over the subscriber loop by exploiting the line transmission technique (e.g. the line coding).

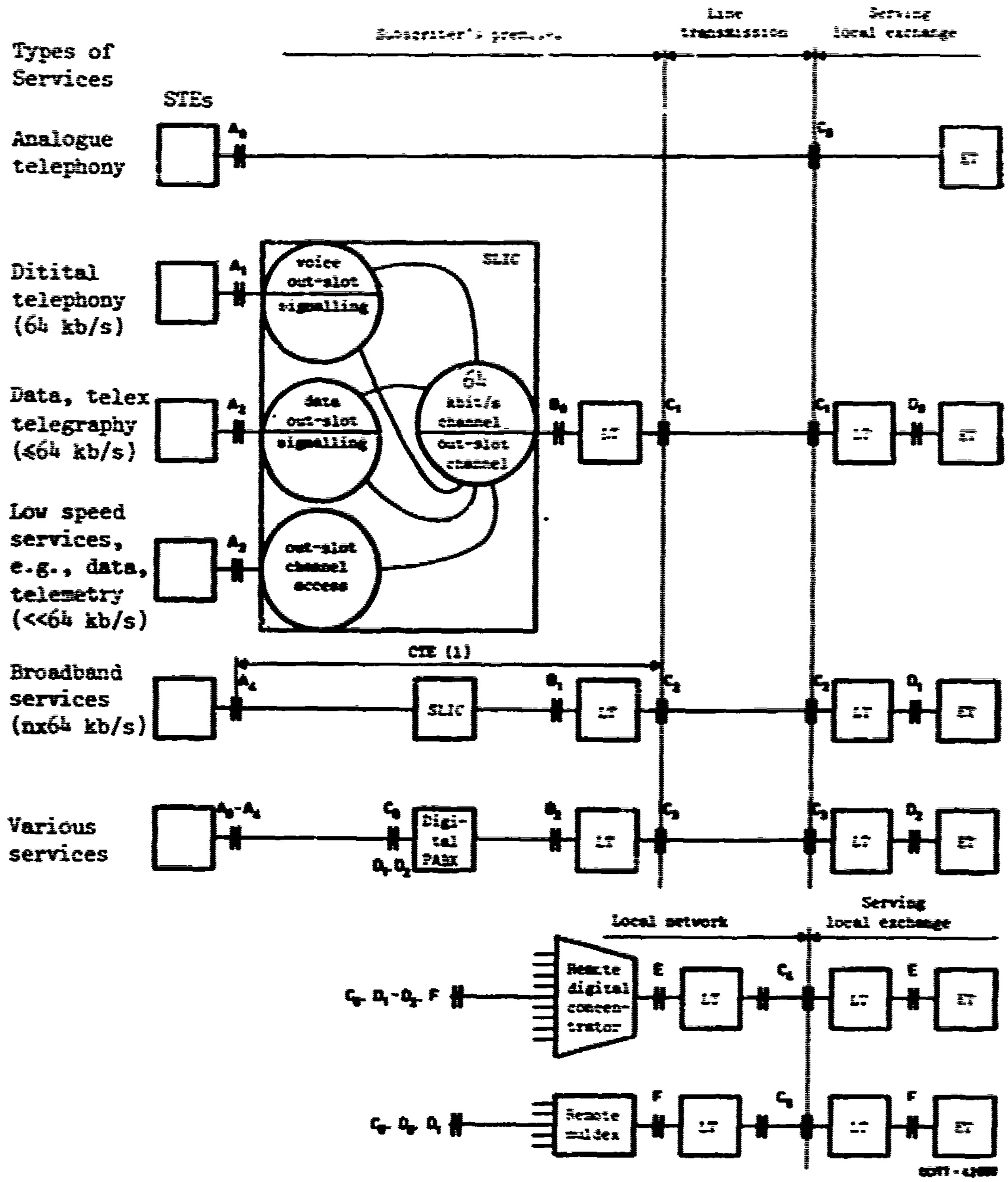
e) Concerning network provided broad-band services requiring n x 64 kbit/s the interfaces B_1, C_2 and D_1 will, depending on the value of n and the conditions in each particular case, be:

- n parallel B_0, C_1 and D_0 interfaces, respectively, or
- C_5 and F interfaces, respectively.

See also 1.4.3.

f) Concerning PABXs (of type 2), interfaces B_0, C_3 and D_2 will match the E, F, C_4 and C_5 respectively. See also 1.4.4.

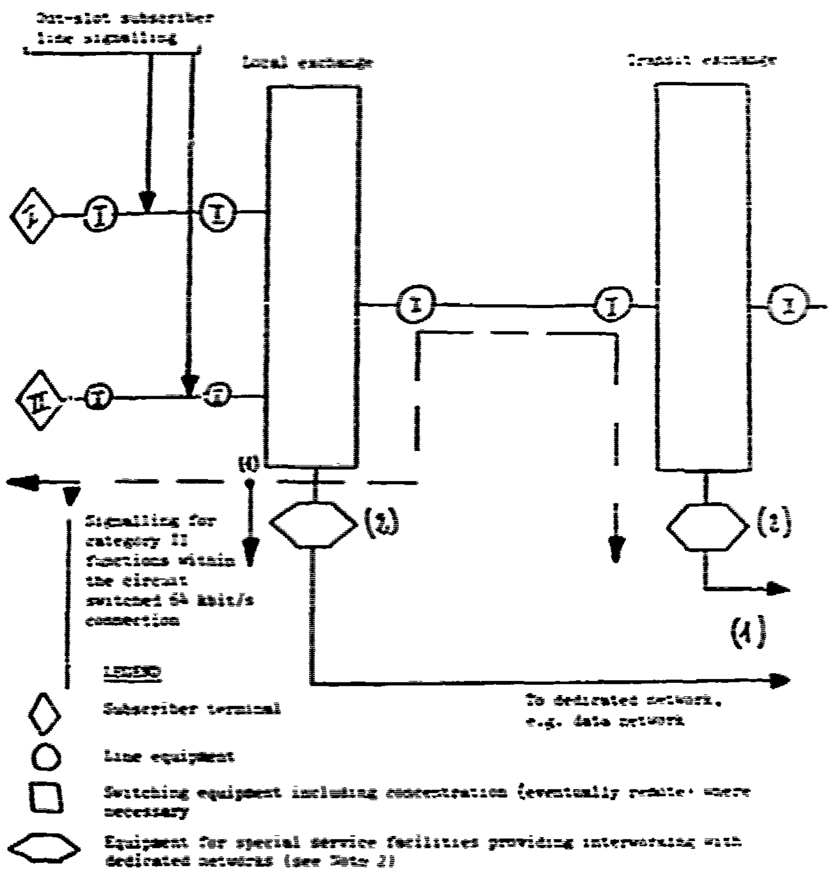
g) Concerning remote digital concentrators and muldexes the use of the standard 2 Mbit/s and 8 Mbit/s interfaces is envisaged. Interfaces E and F should be in accordance with CCITT Rec. G703. Although some differences probably cannot be avoided, interfaces E and F should be based on Recs. G 734 and G. 746. A new muldex with lower capacity than 2 Mbit/s, of which possible standardisation is under study, can also be used for interfaces E and F.



STE = Subscriber Terminal Equipment
SLIC = Subscriber Line Interface Circuit; Function : signal arrangement for line transmission
LT = Line Terminal; Function : line coding, feeding, timing and possibly alignment
CTE = Circuit Terminating Equipment
ET = Exchange Terminal

(1) SLIC and LT-functions may be combined into a CTE

Figure 1 - Possible functional interfaces in local networks for ISDN



CATEGORY 1 : Basic functions for circuit switched 64 kbit/s digital connection, typically for telephony and possibly for circuit switched data. These control and signalling functions are provided in all terminals and exchanges.

CATEGORY 2 : Additional special service control and signalling functions are provided in special terminals and equipments.

Note 1 : The access to dedicated networks may be provided at the local exchange (including possible remote concentration stages) or at the transit exchange.

Note 2 : Connections between subscribers belonging to the same geographical area (category I to category II, probably also category II to category II) may be served by the equipment for special service facilities without passing through the dedicated network.

Figure 2 - Concept of two categories of networks functions

__QUESTION [illegible]/XVIII__ - Customer/network interface
(New Question)

1. What are the principles which should form the basis for detailed study of the customer/network interface to ISDN, taking into account existing and new, analogue and digital, voice and non-voice, service dedicated and multi-purpose terminals ? (To include single access, multi-line access, multi-service access and PABXs.

2. __Considering__

2.1 that the functional requirements for the interface between dedicated-service customer terminals and the ISDN will be identified by Study Groups as follows :

- a) data terminals - Study Group VII
- b) digital telephones - Study Group XII
- c) analogue telephones - Study Group XI
- d) facsimile terminals - Study Group XIV
- e) Teletex terminals - Study Group VIII
- f) voice band data terminals - Study Group XVII

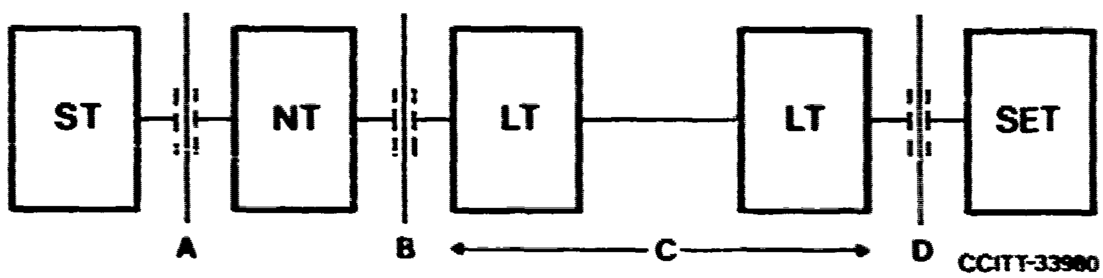
2.2 that some customers will have multi-service terminals using one or more access paths to the ISDN;

2.3 that there will be a need to connect multiplexed groups of digital PABX lines to local digital exchanges in the ISDN;

2.4 that the multiplex arrangements in (2.3) may be applicable to remote multiplexes used to connect several direct lines to the local digital exchange;

2.5 that the principles relating to the customer/network interface, and the associated customer network signalling, will be studied by Study Group XVIII (Question 9/XVIII);

- what are the preferred characteristics to be recommended for the interface between customer terminals and the ISDN (Interfaces A and B on Figure 1).



ST - Subscriber Terminals
NT - Network Termination
LT - Line Termination
SET - Subscriber Exchange Terminal
- Functional Interfaces

Figure 1 - Possible Functional Interfaces for Digital Local Access

QUESTION 3/XVIII - Synchronization in digital networks

(continuation of Question 3/XVIII, studied in 1977-1980)

Considering

a) that general agreement has been reached in the previous study period on the need for high quality timing control in the national digital networks,

b) that the decision to operate international connections in the first instance on a plesiochronous basis has been made,

c) that the interest has been expressed by some Administrations in continuing to study the possibility in the longer term of a synchronized international network,

d) that the requirements on the interconnection of international digital links are defined in Recommendation G.811,

e) that basic agreement has been reached on the need for end-to-end performance requirements on digital connections,

f) that some of these requirements have been defined in draft Recommendation G.822,

g) that further refinements of these Recommendations are required,

1) What are the requirements for synchronization methods in national networks, to provide compatibility in international connections and to ensure the end-to-end performance requirements are met ?

2) What methods should be used for the synchronization of the international ISDN and dedicated IDNs ?

3) What additional slip criteria are to be recommended for the end-to-end performance of a digital connection, during periods of temporary loss of synchronization ?

4) How should the reliability of network clocks be defined and what values should be allocated to reliability ?

5) What synchronization requirements are needed when satellite switched/TDMA mode links are incorporated into the network ?

6) What further studies are required to define the limits of wander likely to be encountered ? How should this be specified ?

7) What modifications or additions are required to Recommendation G.811 (Performance of clocks suitable for plesiochronous operation of international digita links) and G.822 (Controlled slip rate objectives on an international digital connection) ? It will be desirable to harmonize the relevant G, Q and X series Recommendations.

8) What additional Recommendations need to be formulated to satisfy the above points ?

Annex

(to Question 3/XVIII)

Proposed draft Recommendation G.8YY - Jitter and wander in a digital network

(contribution by Nippon Telegraph & Telephone Public Corporation)

1. Introduction

Study Group XVIII agreed to define the Recommendation G8YY specifying maximum tolerable wander as a design object of a frame aligner at a network node. The general guideline, as follows, was established at the May, 1979 meeting.

a) It should be recognized that excess buffer size will significantly increase the delay through the exchange.

b) The specification on wander should be taken as a common design requirement for a frame aligner.

c) In the study of wander, it is necessary to refer to present CCITT Recommendations.

- G.721: hypothetical reference digital path

- G.104: hypothetical reference connection

- G.811: international plesiochronous links

This contribution provides the basis for technical considerations on this matter as well as the text of draft Recommendation G.8YY.

2. Parameter related to wander

2.1 Network configuration

a) The national network has five hierarchical levels.

b) Maximum length of digital links:

long-haul link: 5000 km

short-haul link: 100 km

2.2 Source of wander

Two factors are considered:

a) delay deviation due to temperature variation in transmission facilities;

b) phase deviation of oscillators at a network node.

2.3 Wander due to transmission facilities

2.3.1 Types of transmission systems

To specify the wander of transmission facilities, it is sufficient to take into account only the cable transmission systems. It is because wander via radio-relay systems can be considered to be smaller.

Wander of satellite and optical fiber links is for further study.

2.3.2 Delay variation performances

Two factors should be taken into account to specify the wander value of transmission facilities: cables and repeaters.

a) Performance of cables

Delay performances due to temperature variation can be given as follows:

symmetrical pair

paper-insulated: 3.0 ns/km °C

polyethylene-insulated: 0.3 ns/km °C

coaxial cable (2.6/9.5 mm): 0.03 ns/km °C

b) Performance of repeaters

Delay variation performance mainly due to mistuning of a tank circuit can be specified as follows:

$$\Delta\tau/\Delta T = 2/f_0 \text{ (ns/}^\circ\text{C rep)}$$

Where $\Delta\tau$: delay variation

f_0 : clock frequency in MHz

ΔT : temperature variation

2.3.3 Temperature variation

The values of temperature variation are not yet standardized, but the following values are assumed considering the typical conditions.

	Buried	Aerial
Cable	20 °c	80 °c
Repeater	40 °c	90 °c

(peak-to-peak value)

2.3.4 Installation conditions of transmission facilities

The following typical conditions are considered:

long-haul link ------- coaxial cable (100% buried);

short-haul link ------ paper-insulated pair (100% buried)

or polyethlene-insulated (100% aerial).

2.3.5 Transmission path model

As HRX or HRDP, following digital paths are considered:

long-haul digital link: 5000 km;

short-haul digital link: 100 km (= 2 x 50 km).

2.3.6 Wander value

From the conditions described above, the wander value due to transmission facilities is estimated as follows:

	Daily	Daily and yearly
Long-haul	0 µs	3.6 µs
Short-haul	6.2 µs	10.4 µs
Total	6.2 µs	14.0 µs

(peak-to-peak value)

2.4 Wander generated at a network node

2.4.1 In case of synchronous network

In the case that a network is synchronized by a master-slave or a mutual synchronized system, wander generated at a network node is due to phase deviation of a phase-locked oscillator such as:

a) unstability of an oscillator;

b) phase drift of a phase-locked oscillator;

c) phase deviation due to digital control,

Taking into account these factors, 1/2 UI (peak-to-peak value) is considered to be appropriate as a maximum wander value at each network node.

Therefore, wander generated at each network node is given below.

1.544 Mb/s: $\Delta\tau = 0.324$ µs

2.048 Mb/s: $\Delta\tau = 0.244$ µs

2.4.2 In case of plesiochronous operation

In case of plesiochronous operation, wander generated at a network node is due to instability of oscillators and its value is specified as 20 µs in Recommendation G.811. But as some administrations pointed out, its value is too large from the viewpoint of feasibility.

NTT proposed a revised value of 2.5 µs in the accompanied contribution (COM XVIII No. 398).

3. Wander in a digital network

3.1 Wander in a national network

Wander in each digital link and network node accumulates along digital links, and its accumulation law depends upon network synchronization system: master-slave, mutual and plesiochronous sytems. Therefore, it is necessary to estimate the maximum wander in case of each synchronization system.

3.1.1 Master-slave system

In the case that a national digital network is synchronized by master-slave syste the wander accumulates linearly along digital links as shown below:

$\Delta\tau_T = \Sigma\Delta\tau_L + \Sigma\Delta\tau_N$

$\Delta\tau_T$: total wander in a digital network

$\Delta\tau_L$: wander in each digital link due to transmission facilities

$\Delta\tau_N$: wander generated at each network node.

Based upon the network parameters and the wander values in each section as shown above, total wander is given as follows·

$\Delta\tau_T = 14.0 + 0.32 \times 10$

$= 17.2$ (µs)

3.1.2 Mutual synchronization system

In this case, the maximum wander in a national network is given by:

$\Delta\tau_T =$ [wander of one digital link] + [wander generated at both end network nodes]

$= 14.0 + 2 \times 0.32$

$= 14.62$ (µs)

3.1.3 Plesiochronous operation

In this case, the maximum wander is given by:

$\Delta\tau_T$ = [wander of one digital link] + [wander generated by oscillators at both end network nodes]

= 14.0 + 2.5

= 16.5 (μs)

These estimations lead to the conclusion that the maximum wander in a national digital network is 17.2 μsec for any network and any network synchronization system.

3.2 Wander in an international network

To estimate the wander in an international network, the two cases should be considered.

Case 1: Digital path is set up through international gateway exchanges connected by plesiochronous operation.

Case 2: Digital path is set up directly between neighbouring exchanges in neighbouring countries where a national network is synchronized to a reference clock.

In case 1, wander accumulated in each national network is isolated by the international link using plesiochronous operation. Therefore, the maximum wander is equal to that of a national network.

In case 2, the maximum wander is given by

$\Delta\tau$ = [wander of a national network] + [wander generated by reference clocks]

= 17.2 + 2.5

= 19.8 (μs)

4 Jitter in a digital network

Jitter at each hierarchical level is specified in Recommendations for MULDEX and interfaces as shown below:

1.544 Mb/s: Recommendation G.743

2.048 Mb/s: Recommendation G.703

8.448 Mb/s: Recommendation G.703

5 Recommendation of jitter and wander

These discussions lead to the Recommendation G.8YY which specifies jitter and wander in a digital network.

Draft Recommendation G.8YY is attached as an Annex to this contribution.

Annex

(to proposed draft Recommendation G.8YY)

Jitter and wander in a digital network

1 General

This Recommendation deals with the lower limit of maximum tolerable jitter and wander in digital networks to establish common design requirement of a frame aligner at each network node. The frame aligner is located in a digital exchange terminal, a synchronous digital multiplex or other synchronous equipments. Digital networks synchronized by the following systems are assumed:

international links ------- plesiochronous (G.811);

national network ---------- synchronous (master-slave or mutual) or plesiochronous.

2 Jitter and wander in a digital network

2.1 Jitter and wander in a national network or international links via gateway offices

The maximum tolerable jitter and wander at an input of a frame aligner in a national network or an international link via gateway offices are specified by Figure 1 and Table 1.

2.2 Jitter and wander for local neighbourhood international links

The maximum tolerable jitter and wander for local neighbourhood international links are specified by Figure 1 and Table 2.

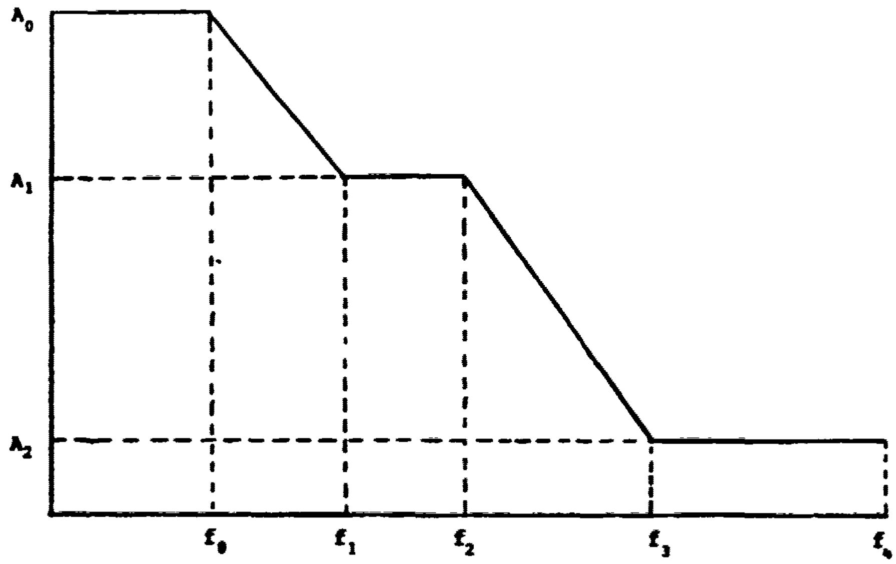


Figure 1 - Mask of tolerable sinusoidal jitter and wander

TABLE 1

Values for the mask of tolerable jitter and wander at frame aligners

	2048 kbit/s	8448 kbit/s	1544 kbit/s
A_0 (us)	17.2	17.2	17.2
A_1 (UI)	1.5	1.5	2.0
A_2 (UI)	0.2	0.2	0.05
f_0 (Hz)	3×10^{-8}	3×10^{-8}	3×10^{-8}
f_1 (Hz)	20	20	10
f_2 (Hz)	2.4×10^3	400	200
f_3 (Hz)	18×10^3	3×10^3	8×10^3
f_4 (Hz)	100×10^3	400×10^3	40×10^3

Note: UI = Unit Interval

for 1544 kbit/s systems 1UI = 648 us

for 2048 kbit/s systems 1UI = 488 us

for 8448 kbit/s systems 1UI = 118 us

TABLE 2

Values for the mask of tolerable jitter and wander at frame aligners

	2048 kbit/s	8448 kbit/s	1544 kbit/s
A_0 (us)	19.8	19.8	19.8
A_1 (UI)	1.5	1.5	2.0
A_2 (UI)	0.2	0.2	0.05
f_0 (Hz)	3×10^{-6}	3×10^{-6}	3×10^{-6}
f_1 (Hz)	20	20	10
f_2 (Hz)	2.4×10^3	400	200
f_3 (Hz)	18×10^3	3×10^3	8×10^3
f_4 (Hz)	100×10^3	400×10^3	40×10^3

QUESTION 4/XVIII - Signalling for the ISDN

(continuation of part of Question 2/XVIII, studied in 1977-1980)

Considering

a) the need for IDNs to be a subset of the ISDN;

b) that Recommendations for Signalling System No. 7 have been prepared in Study Group XI (message transfer part, common to all users and services, and telephony user part) and in Study Group VII (data user part);

c) that Recommendation X.25 exists for recommending an interface between DTE and DCE for terminals operating in the packet mode on public data networks;

d) that Signalling System No. 7 is likely to be used as the basis for network signalling for other services in an ISDN;

e) the need for additional requirements for customer access signalling identified in Study Group XVIII;

f) that interworking between customer access signalling and CCITT Signalling System No. 7 will be required;

g) that functional layering concepts for data networks are being developed in CCITT;

h) that coordination of the various signalling requirements in an ISDN is necessary;

What are the principles which should form the basis for detailed study of signalling for ISDN and provide a framework within which equipment Recommendations can be produced by the specialized Study Groups in regard to :

1. for customer access/network signalling for terminals (including multi-service terminals and PABXs) connected to an ISDN ?

2. for handling inter-exchange signalling for the various services in an ISDN ?

3. for through-signalling from customer equipment to network-provided resources and from customer access including multi-service terminals to another customer equipment, end to end ?

4. for signalling between ISDN exchanges and dedicated networks ?

5. to provide signalling arrangements for the switching in and out of digital processing devices as required in an ISDN ?

6. to provide for the application of the functional layering principles in development of signalling protocols in an ISDN ?

Note : Relevant material is to be found in Annexes 3 and 4 to Question 1/XVIII.

QUESTION 5/XVIII - Switching for the ISDN

(continuation of part of Question 2/XVIII, studied in 1977-1980)

Considering

- the need for IDNs to be a subset of the ISDN,

- that Recommendations for digital transit exchanges have been prepared in Study Group XI,

- the need for exchanges to be able to switch 64 kbit/s circuits which may be carrying voice, data or other services,

- the need for exchanges to meet operational and performance requirements of voice, data and other services,

- the need to control processing devices which may be included in some connections,

- the need to provide interworking and interconnection between different standards and different types of dedicated networks,

- the contribution of exchanges to overall network performances,

- the need to provide interfaces appropriate to equipment which will be connected to exchanges, from the customer side and from the interexchange side,

"What are the principles which should form the basis for detailed study of switching for ISDN, and provide a framework within which equipment Recommendations can be produced by the specialised study groups. The following points should be included in the studies :

1. The interface between local network digital transmission and the local exchange (or concentrator).

2. The interface between trunk and junction digital transmission systems and local and trunk exchanges of ISDN.

3. The interface between ISDN exchanges and transmission systems to dedicated networks.

4. Provision of general features required by many services on ISDN.

5. Provision of service-dependent features required in parts of ISDN.

6. Method of controlling signal processing devices (e.g. echo suppressors).

7. Arrangements for deriving charging information according to the requirements of different services carried on ISDN.

8. Arrangements for the use, identification and analysis of service indication.

9. Network addressing and numbering and routing options for interworking between the ISDN and other networks, and interconnexions of customers connected to the ISDN customers connected to other networks.

10. Methods of carrying wider-band services by means of multi-slot connexions.

11. Switching programme requirements consistent with network requirements (in cooperation with Question 9/XVIII).

12. Operation and maintenance of switching (in cooperation with Question 12/XVIII).

13. Any additional provisions to facilitate a smooth transition towards a comprehensive ISDN".

Note : Relevant material is to be found in Annexes 3 and 4 to Question 1/XVIII.

QUESTION 6/XVIII - Definition for digital networks

(continuation of Question 7/XVIII, studied in 1977-1980)

What definition should be given to terms used for digital systems (including switching, signalling, synchronization and transmission systems) which form part of digital networks ?

Note 1 : Studies on this Question should be based on Recommendation G.702.

Note 2 : The Rapporteur for this Question will act as coordinator for the study of related definitions produced in other Study Groups, e.g. Study Groups VII and XI.

Note 3 : It is understood that in the present set of Questions the term "Digital Networks" includes both the IDN and the ISDN.

QUESTION 7/XVIII - Encoding of speech and voice-band signals using methods other than PCM, in accordance with Recommendation G.711

(continuation of Question 10/XVIII, studied in 1977-1980)

Considering that

a) PCM encoding for speech is now widely used in telecommunication networks for telephony (A-law and μ-law),

b) progress in technology is likely to offer other coding methods which are of a technical and economic interest,

c) there is a desire to make efficient use of transmission paths,

d) proliferation of standards in the world wide digital network leads to the need for code converters and to operational complications,

e) the IDN and ISDN studies are based exclusively on 64 kbit/s paths

the following should be studied :

Should any other encoding method (e.g. adaptive differential PCM, delta modulation or sub-band coding) be recommended for use in international networks ? If so, for what applications are they suitable and what are the relevant transmission performance criteria ? What are the inter-working problems with systems using standard PCM encoding ?

In the frame of this Question the following problems should be considered :

1. choice of encoding methods,
2. choice of a bit rate for encoding of speech signals,
3. effect of digital network impairment on voice and voice-band data signals transmitted over codecs (ADPCM, DM and others),
4. voice-band data transmission at various rates in channels formed by ADPCM, DM and other methods,
5. transmission performance of codecs connected in tandem (ADPCM, DM and others),
6. compatibility with digital networks based on 64 kbit/s time slots,
7. compatibility with digital signals encoded by other methods, for example 64 kbit/s PCM,
8. the performance of the lower rate speech coding schemes when they are connected in tandem with an appropriate digital conversion to interface with the existing µ-law and A-law 64 kbit/s Recommendation G.711,
9. quantification of the impairment to voice and in-band data services when the digital path produced errors in the range allowed by Recommendation G.821,
10. the methods of assessment used in the evaluation of speech performance (comments of Study Group XII will be requested on this matter),
11. usage of encoding methods other than PCM for wideband speech signal transmission,
12. the effect of reducing the number of code words used to eliminate the all-zero code word in order to facilitate world standardization.

Annex 1

(to Question 7/XVIII)

Reply to Question 10/XVIII (Other methods of encoding than PCM), study period 1977-1980

The studying being carried out during the last study period 1977-1980 showed that problems of choice of a differential encoding method and of an encoding bit rate are most important and urgent.

Many Administrations endorsed the opinion that it would be highly undesirable to have a situation which in the past resulted in adoption of two encoding PCM laws (A and μ) and which, subsequently, necessitated the A to μ conversion and vice versa.

The majority of Administrations agreed that when using differential methods of encoding the encoding bit rate should be submultiple to 64 kbit/s and multiple to 8 kbit/s. The preferable encoding bit rate is 32 kbit/s.

During the preceding study period 1973-1976 the study mainly related to adaptive delta modulation. The present study period has shown greater interest in adaptive differential PCM (ADPCM) at 32 kbit/s for voice and voice-band data signal transmission.

Study Group XVIII takes note of the results obtained by Study Group XII for an estimate of a value of quantizing distortion and number of impairment units with 32 kbit/s ADPCM process.

Moreover, the opinion was expressed that the differential encoding methods could be reasonably used not only for reducing a bit rate for telephony but also for transmitting wideband voice signals without increasing the bit rate of 64 kbit/s.

An opinion was also expressed that when choosing the bit rate the methods and laws of encoding the problems of integration of services in digital networks should be taken into consideration along with the impact of the introduction of this new encoding method into integrated networks. Moreover, the attention was drawn to necessity of studying the problem of compatibility with digital signals encoded by other methods, for example 64 kbit/s resulting from PCM encoding in accordance with Recommendation G.711.

The results obtained during the present study period gave no opportunity to prepare any draft Recommendations on other methods of encoding than PCM. It should be noted that the most pressing problem is that of choosing 32 kbit/s as a bit rate for differential encoding methods.

Annex 2

(to Question 7/XVIII)

Telephone signal encoding at 32 kbit/s : Concept of non-accumulation of impairments in successive code conversions

(contribution from the French Administration)

1. Introduction

A distinction should be drawn between the two types of equipment indicated below when considering an encoding method other than PCM, such as PCM differential adaptive encoding at 32 kbit/s (PCMDA) :

- the terminal which converts the AF signal directly into an encoded signal, e.g. at 32 kbit/s, and vice versa;
- the code converter which converts the encoded PCM signal at 64 kbit/s (A-law or μ-law) into a signal encoded at a lower bit rate, e.g. 32 kbit/s, and vice versa.

"Encoding" is the operation effected by a terminal and "code conversion" that effected by a code converter.

A given encoding technique will therefore have the property of non-accumulation of impairments in successive code conversions, if the following operations :

AF ⟶ PCM ⟶ PCMDA ⟶ PCM ⟶ AF

and AF ⟶ PCM ⟶ PCMDA ⟶ PCM ⟶ PCMDA ⟶ PCM ⟶ AF

produce the same impairment regardless of the number of intermediate

PCM —— PCMDA —— PCM sequences.

This property relates solely to successive code conversions.

Paragraph 2 shows the advantage of this property for the introduction of 32 kbit/s encoding techniques into a digital network.

The French Administration has studied a differential 32 kbit/s PCM coding technique with adaptive quantizer and fixed predictor which possesses this property (see COM XVIII-No. 248, study period 1977-1980).

To improve the subjective quality of 32 kbit/s PCMDA encoding techniques using a fixed predictor, some Administrations (including the French) have proposed the use of 32 kbit/s adaptive quantizer and adaptive predictor differential PCM encoding techniques. The French Administration would also point out to other Administrations studying such encoding techniques that it would be useful if these techniques also possess the property of non-accumulation of impairments in successive code conversions. Studies along these lines are at present in progress in France.

2. Non-accumulation of impairments in successive code conversions

The purpose of this paragraph is to consider the repercussions of the property of non-accumulation of impairment on the successive code conversions which can appear in a digital network.

Let us consider a digital network with TDM switching and a bit rate of 64 kbit/s, and the case (Figure 1) of a call routing following several digital transmission paths separated by TDM switching centres. A priori, if we use a conventional 32 kbit/s coding in transmission and we put a code converter at each switching/transmission interface, each code converter will introduce its own impairment and the total impairment will be proportional to the number of circuits involved.

If, on the other hand, the 32 kbit/s coding used for transmission does not accumulate impairments in successive code conversions, the total impairment will correspond to the impairment caused by a 32 kbit/s coding/decoding irrespective of the number of circuits used for the call.

This type of digital routing will become very common once network digitization is sufficiently advanced, hence the importance of impairment non-accumulation in successive code conversions.

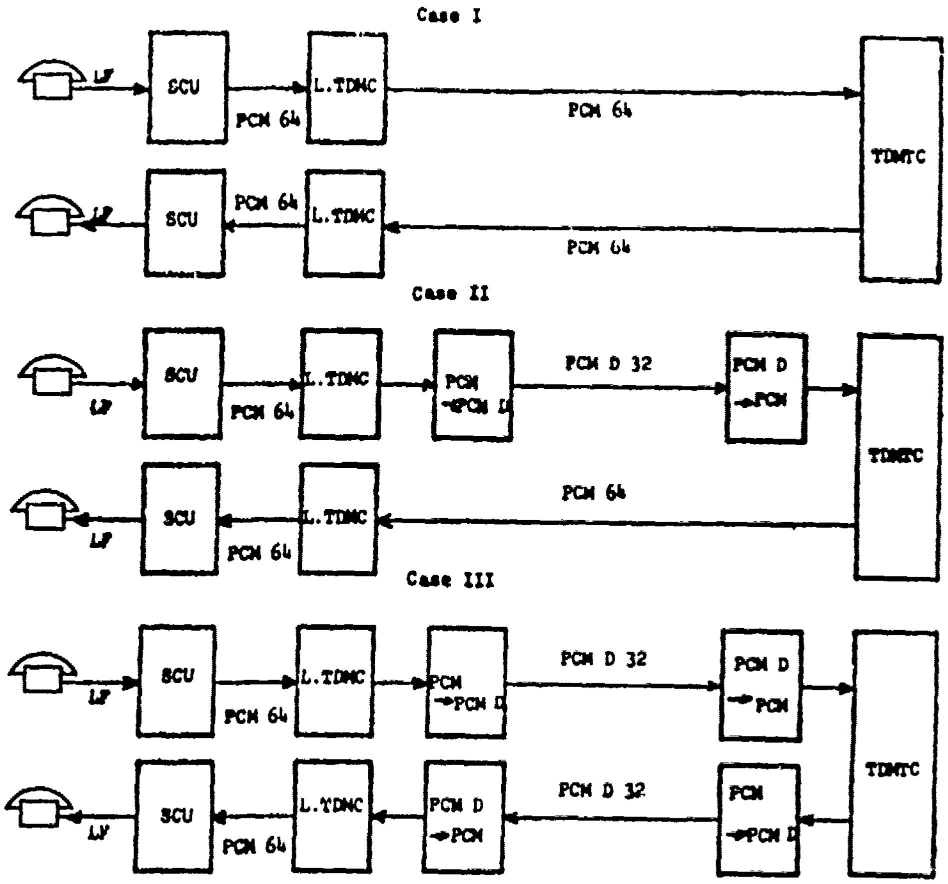


Figure 1 - Examples of digital calls

- Case I — Example of 64 kbit/s digital call
- Case II — Example of digital call using a 32 kbit/s section : the impairment compared with Case I is that of a PCM D coding/decoding
- Case III — Example of digital call using <u>two</u> 32 kbit/s sections

The impairment is the same as in Case II provided that the 32 kbit/s coding has the property of non-accumulation of impairment in the successive code conversions

SCU : Subscriber connection unit

LTDMC : Local TDM centre

TDMTC : TDM transit centre

PCM PCM D Code converter PCM PCM D

PCM D PCM Code converter PCM D PCM

Only one call direction is shown

Annex 3

(to Question 7/XVIII)

Quality evaluation of codecs at 32 kbit/s

(contribution by Bell-Northern Research)

1. Introduction

Recent experimental work on new ADPCM Codec designs conducted at Bell-Northern Research indicate that the speech quality obtained with 32 kbit/s ADPCM comes close to that obtained with 8-bit/μ=255 companded PCM codecs. In particular, two codec designs were evaluated, one employing a fixed third-order predictor and an adaptive non uniform quantiser, hereafter referred to as the 'F32' codec, the second employing an adaptive third-order predictor and the same quantiser as before, hereafter referred to as the 'V32' codec.

2. Results

(a) Both ADPCM codecs meet the signal-to-total distortion ratio template of CCITT Recommendation G.712, Figure 5 for frequencies below 1100 Hz. For higher frequencies, the signal-to-total distortion ratio drops rapidly for the 'F32' codec, but remains above 30 dB for the 'V32'.

(b) The subjective speech preference tests show that the quality of speech transmitted over the 'V32' codec is comparable to that of speech transmitted over an 8-bit, μ=255 PCM codec. The quality of speech transmitted over the 'F32' codec is slightly worse than that of speech transmitted over an 8-bit but better than that over a 7-bit, μ=255 PCM codec.

(c) Voice-band data transmission at data rates up to 4800 bits over the 'V32' codec resulted in bit error ratios of less than 10^{-6}. In contract, the 'F32' codec shows bit error ratios of more than 10^{-5} at the tested data rates of 2400 bits and 4800 bits.

(d) The test results confirmed that signal-to-total distortion ratio with speech signal input is not suitable to accurately assess the quality of ADPCM codecs. For the same subjective quality, the signal-to-total distortion ratio (speech input signal) of the tested ADPCM codecs is numerically approximately 8 dB lower than the corresponding signal-to-total distortion ratio of PCM codecs.

A modified segmented signal-to-total distortion ratio (speech input) quality indicator modelled after Richards [1] was also tested and showed no such bias. The quality indicator Q is defined here as

$$Q = 20 \log q$$

where $\log(1+q^2) = \frac{1}{N}\sum_{n=1}^{N} \log\left[1+\sum_{i=1}^{M} x_i^2 \Big/ \sum_{i=1}^{M}(x_i-\hat{x}_i)^2\right]$

and M = 64 samples at 8 kHz

N = number of 8 ms blocks in the signal

x_i = original speech samples

$\hat{x}_i$ = reconstructed speech samples

(22)

3. **Conclusions**

A single 32 kbit/s ADPCM encoder appears to provide adequate performance for voice transmission over an error free channel.

(a) For voice transmission only ADPCM with fixed prediction (F32) is sufficient.

(b) If both, voice and voice-band data signals are to be transmitted, ADPCM with fixed prediction (F32) is not adequate but the ADPCM with adaptive prediction (V32) may be suitable.

(c) To ensure speech quality better than that of 7-bit, μ=255 companded PCM, at least third-order prediction in ADPCM codecs is indicated.

REFERENCE

1. RICHARDS, D.L. Speech-transmission performance of PCM systems, Electronics Letters, 1, pp. 40-41, 1965.

Annex 4

(to Question 7/XVIII)

Speech coding at 32 kbit/s

(Extract from COM XVIII-No. 185 - Italy)

1. Introduction

This Annex describes a digital Differential Pulse Code Modulation with adaptive quantization (ADPCM) developed and tested in Italy. Its performances in terms of objective as well as subjective measurements are shown and compared with a 64 kbit/s PCM system.

2. ADPCM 730 : A fully digital adaptive differential PCM

Figure 1 shows the block diagram of a DPCM with adaptive quantization.

The input signal digital samples s_n are compared with the sample $\hat{s}_n$, being the prediction filter output; this difference e_n, called prediction error, is coded and transmitted.

Prediction is performed by using Equation (1) :

$$\hat{s}_n = r_{n-1} - 0.5\, r_{n-2}$$
$$r_n = d_n + \hat{s}_n \qquad (1)$$

The predictor coefficients were computed so as to minimize the mean square error e^2_n. A filter with two coefficients only was chosen, because by increasing the number of coefficients the signal-to-prediction error ratio does not improve considerably.

The adapter modifies the 4 bit quantizer step-size on the basis of a short-time estimation of the r.m.s. of signal e_n to be transmitted. In order to avoid the transmission of the step-size, it is computed both at the transmitter and at the receiver using samples d_n, i.e. the prediction error after quantizing and inverse quantizing.

Step-size computation is performed by using the following equations :

$$Q_n = c\,P_n \qquad (2)$$

$$P_n = (1 - 2^{-k})\,P_{n-1} + 2^{-k}\,d_n \qquad (3)$$

where c = 0.5 and k = 2 were chosen to maximize the signal-to-noise ratio measured by computer simulations.

3. <u>ADPCM 730 : Objective and subjective performances</u>

The signal-to-noise ratio is computed as :

$$S/N = 10 \text{ Log } \frac{< s^2{}_n >}{< (s_n - r')^2 >}$$

where s_n and r' are referred to in Figure 1.

S/N consists of the two following terms : S/E, the improvement due to prediction and E/N, the signal-to-noise ratio of the quantizer.

In the case of the predictor, with the two chosen coefficients, S/E is slightly higher than 4 dB, while E/N is about 17 dB; then S/N is greater than 21 dB.

Figure 2 shows the objective performances of the system on the whole telephone signals range, in terms of S/N ratios of a 64 kbit/s and of a 32 kbit/s log - PCM for comparison reasons.

It has to be pointed out that S/N is always greater than 20 dB. In fact, it was verified that if S/N exceeds this value, for multiplicative noise and telephone speech signals, subjective quality is quite independent from the same S/N value.

In ADPCM 730 the noise is really multiplicative, as shown in Figure 3, owing to the quantizer step-size adaptation. In fact, the noise is always masked by the speech signal and then it is less perceivable than an additive one.

Moreover, ADPCM can present an adaptive quantizer characteristic saturation which can be referred to as slope-overload, because the system is differential and which is due to the deconvolution process. An example is shown in Figure 4, where a typical speech waveform and noise are plotted.

This kind of noise does not present a great infuence on subjective assessments, while it is more important, for the overall noise objective measurement, starting from the given definition.

In the case of the ADPCM 730 system S/N, evaluated by excluding the samples presenting slope-overload, is 2 to 3 dB greater than overall S/N.

Subjective tests were performed by the categorical judgment method for 64 kbit/s log - PCM channels as well as for the described ADPCM system.

The tests were carried out during a conversation between two operators who gave their judgment.

The mean opinion was obtained from 20 people assessments and every one carried out two tests under each measurement condition.

Figure 5 shows the mean opinion score versus the link overall reference equivalent for 64 kbit/s log - PCM and 32 kbit/s ADPCM channels. It can be seen that the mean opinion score presents only a small difference in both conditions.

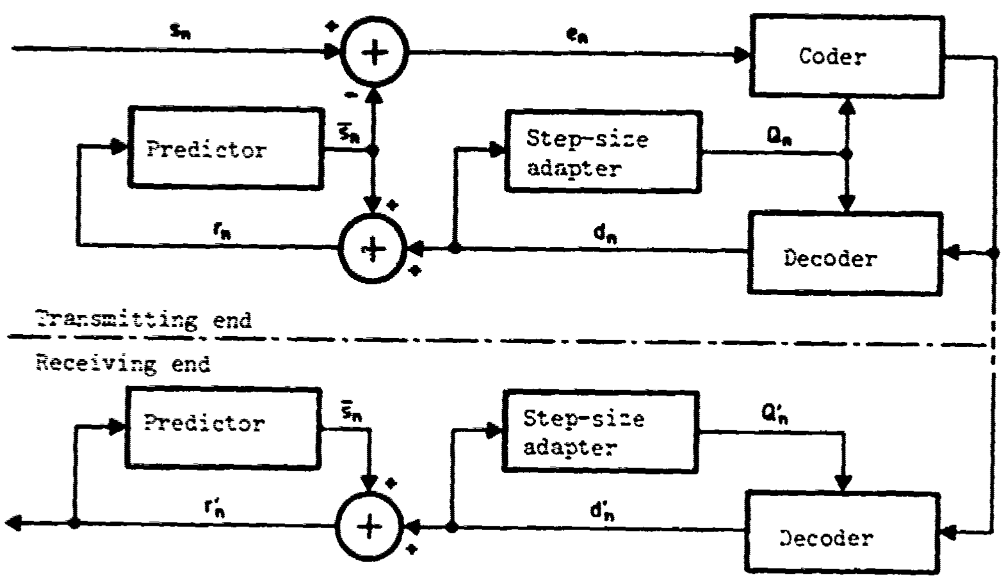
Tests have also been carried out on the transmission of data signals, telex and MF signalling in the voice bandwidth.

For telex and MF signalling, no impairments due to the ADPCM coding process have been observed.

As regards data signals, no quality impairments are introduced by the experimental equipments when the data bit rate is lower or equal to 2400 bit/s and no more than two A/D - D/A conversions in tandem are included. Data transmission tests at 4800 bit/s were not completely satisfactory. Studies are being carried out for assuring a good transmission quality also at this bit rate.

It is worth noticing that in the equipments developed, an algorithm is provided in order to prevent the propagation of errors introduced by the channel. Therefore, the quality of the digital connection in terms of error rate is not impaired by the ADPCM coding process.

Listening tests showed that quality is not considerably influenced by channel error probabilities up to 10^{-2}.



CCITT - 47120

Figure 1 - DPCM system with adaptive quantization

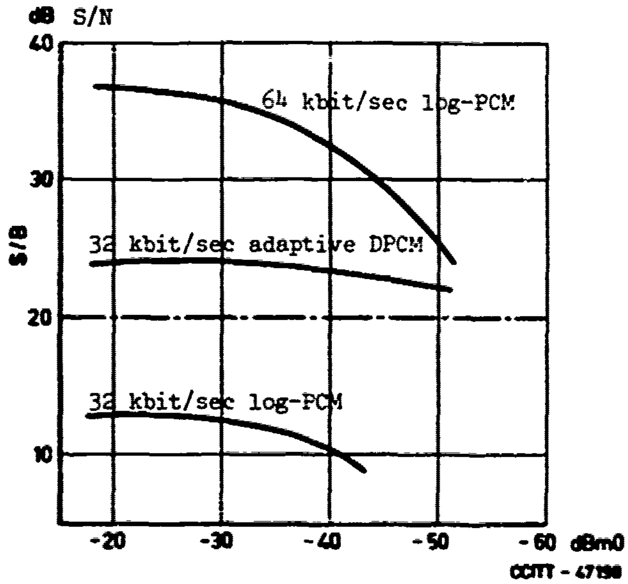


Figure 2 - S/N performance of adaptive DPCM and log-PCM versus input speech level

Figure 3 - Speech waveform a) and multiplicative noise b) due to the adaptive step-size quantizer encoding. The waveform b) is multiplied by a factor 10 in comparison with a)

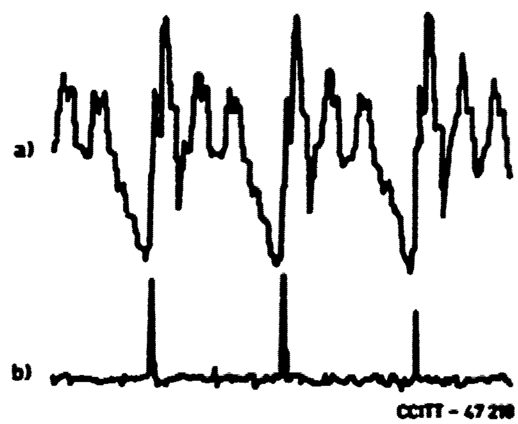


Figure 4 - Typical waveforms in a DPCM system with adaptive quantizer :
a) input signal,
b) quantizing noise.

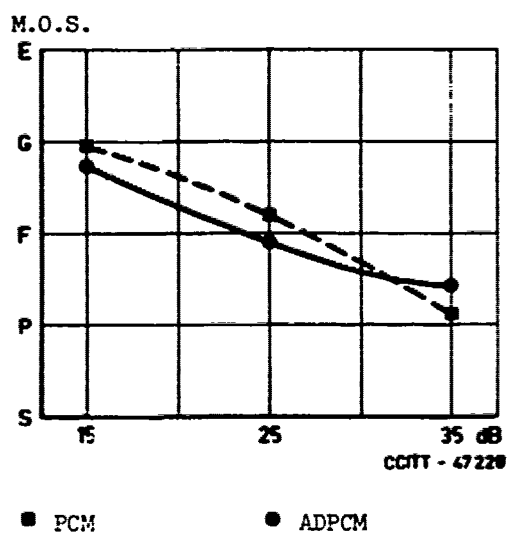


Figure 5 - Mean opinion score for PCM and ADPCM vs. reference equivalent

Annex 5

(to Question 7/XVIII)

Use of differential encoding for "wideband speech" in the ISDN

(contribution from Federal Republic of Germany)

Differential encoding, such as delta modulation or ADPCM, has been taken into consideration in the interest of reducing the bit rate required for voice transmission. For applications of this kind, the majority of Administrations agreed that the encoding bit rate should be 32 kbit/s (see Reply to Q.10/XVIII, Study Period 1977-1980).

This Contribution is intended to draw the attention of S.G. XVIII to the fact that differential encoding may also be used for an entirely different purpose, namely for providing a "wideband speech" service in the Integrated Services Digital Network (ISDN). A suitable (and rather simple, hence economically attractive) encoding scheme is the well-known Continuously Variable Slope Delta Modulation (CVSD) [1] which, when used at 64 kbit/s, enables speech with a bandwidth of about 6 to 7 kHz to be encoded with good quality. This could be attractive, especially in connection with "loudspeaker telephones".

Compared with the encoding of wideband speech by PCM at 128 kbit/s, CVSD has the advantage that only 64 kbit/s are used (no multi-slot service with accompanying problems of providing "digit sequence integrity" and of increased blocking probability).

An additional advantage may be seen in the fact that, if a uniform encoding scheme is standardized, no code conversion in international connections is required, in contrast to PCM encoding.

In the ISDN, it may further be attractive to provide a combined voice/non-voice service where e.g. 48 or 56 kbit/s are used for speech encoding and e.g. up to 8 kbit/s for data or other purposes.

Reference

[1] Hosokawa, S.; Yamashita, K.: Companded delta modulation coders of the 1/2 power and 2/3 power types. Electronics and Communications in Japan, Vol.51-A (1968), No.11, pp. 18-26.

QUESTION 8/XVIII - Digital speech interpolation systems

Considering

a) that digital speech interpolation systems may be used in transmission systems of both national and international networks in the future;

b) that speech interpolation systems may adversely affect the operation of other equipment such as echo cancellors, automatic equalizer;

decides that the following should be studied :

1. what should be the characteristics of digital speech interpolation (DSI) systems ?

2. are special precautions necessary for the combined usage of digital speech interpolation devices with other digital processors such as echo controlloing devices ?

Note : Network implications of digital speech interpolation systems are to be studied under Question 1/XVIII and the performance of connections over digital speech interpolation systems under Question 9/XVIII.

QUESTION 9/XVIII - General network performance aspects of integrated digital networks

(continuation of part of Question 1/XVIII, studied in 1977-1980)

This Question is concerned with studies related to the general performance of an ISDN capable of satisfying the requirements of many different services.

Account will be taken of the performance requirements being established by other study groups in CCITT and CCIR for telephony and other services. Study Group XVIII will keep these Study Groups XI, VII, XV and CMTT informed of the likely performance of common digital building blocks.

In studying this Question reference should be made to draft Recommendation G.102 which relates to transmission performance objectives and Recommendations.

Point A. Transmission performance of digital networks

Considering

a) that some parameters for the transmission performance for the IDN for telephony have already been established,

b) that further types of impairments need to be taken into account,

c) that certain impairment causes are interdependent,

d) that design objectives are urgently required for circuits and systems,

What are the network transmission performance and equipment design objectives necessary for the IDN, the ISDN and the appropriate evolution towards the ISDN ?

The points listed below require particular attention in the studies, whereby account should be taken of all likely media that might be utilized. These include radio relay, satellite, metallic and optical fibre cables. Close collaboration will be required with other CCITT and CCIR study groups.

1. What amendments or additions are necessary to the network performance objectives for errors and slips as recommended in Recommendations G.821 and G.822 respectively ? (See Annexes 1, 2 and 3 and Question 3/XVIII)

2. Which additional types of impairments and characteristics should be studied, and what recommendations should be made in terms of network transmission performance objectives (e.g., jitter, wander, short interruptions and transmission delay) ? (See Annex 4)

Particular attention should be given to the interdependence of certain performance objectives.

3. Are the hypothetical reference connections recommended in G.104 suitable for the study of network transmission performance objectives ? What amendments or additions are necessary ?

Note : They should be brought in line with Recommendation G.103. Recommendation X.92 for data networks should be considered.

4. How should the overall network transmission performance objectives be apportioned to the individual component parts making up the connections ?

5. What approach should be adopted in converting transmission performance objectives for individual items into equipment design objectives and commissioning objectives ? (See Annexes 5, 6 and 7).

6. Are the hypothetical reference digital paths defined in G.721 a suitable tool for defining the transmission performance of network component parts and the systems design objectives ? What amendments are required ? What additional HRDPs should be recommended ?

7. What is the impact of digital signal processing devices (e.g. digital pads, echo controllers, encoding law converters, digital speech interpolation devices, etc.), on network transmission performance, and what is their dependence on network transmission performance and traffic conditions ? What guidelines should be established for their use ? (See Annex 8)

In considering the introduction of digital pads and other digital processes, particular reference should be made to Recommendation G.121 as amended at Geneva 1980 and Recommendation G.142 as amended at Geneva 1980.

Note 1 : Study Group XVI are studying the network planning rules concerning the transmission performance effects of some of the above-mentioned devices, which will be used in mixed analogue/digital networks. Account should be taken of these studies.

Note 2 : Supplement No. 21 (Canada : BNR) gives values for quantizing distortion units for various digital processing devices.

8. What transmission performance parameters should be defined to ensure appropriate evolution towards the ISDN and what values should be recommended for each parameter ?

Point B. Call processing performance of digital networks

Considering

a) that equipment can be designed to measurable performance standards,

b) that networks can be implemented to controllable performance standards,

c) that there is an optimum balance between cost, technology, and the needs and expectations of customers,

d) that some parameters for the quality of service for the IDN for telephony have already been established,

1. What are the call processing performance and design objectives for the IDN, the ISDN, and the appropriate evolution towards the ISDN ?

In particular, performance parameters and their values should be defined on an overall basis (customer to customer) and values apportioned as appropriate to nodes and links in the network, such as call set-up delays, call failures caused by congestion, call failures caused by equipment malfunction and loss of service (availability).

2. What methods should be used to measure the call processing performance ?

3. Which hypothetical reference models should be used for call processing performance determination ?

Note : The study of this Question will take account of consideration by CCIR Study Group 4 (see Contribution COM XVIII-No. 6).

Annex 1

(to Question 9/XVIII)

Considerations on the relationship between mean bit error ratio, averaging periods, percentage of time and percent error free seconds

(Contribution by Bell-Northern Research)

Recommendation G.821 specifies an averaging period of to = 1 minute to determine mean bit error ratio (BER). Simultaneously, error performance is also specified in term of % error free seconds (EFS). Both performance objectives are given for a digital connection at 64 kbit/s.

This Annex offers additional considerations regarding the relationship between Mean Ber's with various averaging periods, percentage of time that the Mean Ber is better than 1×10^{-6} and % EFS. The calculations are made on the basis of an assumed Poisson distribution of bit errors on a 64 kbit/s digital connection.

The relationship between the long term Mean BER and % EFS can be shown to be

$$\% \text{ EFS} = 100e^{-BE}$$

where

e = base of natural logarithm

B = Bit rate in bits/s

E = Long Term Mean Error Probability ($n \times 10^{-x}$), equivalent to a long term Mean BER of $n \times 10^{-x}$

This relationship is shown in Figure 1. Long term Mean Error Probability in this context implies Bit Errors averaged over a sufficiently long period to yield a constant BER.

Recommendation G.821 specifies in Table 1 a threshold between acceptable and degraded performance of a Mean Ber = 1×10^{-6} averaged over To = 1 minute. In order to meet this criterion on a 64 kbit/s digital connection 3 or less errors must be counted. For randomly (Poisson) distributed errors the probability P of this occuring is

$$P(n \leq 3) = \sum_{0}^{n=3} \frac{(60\ BE)^n}{n!} e^{-(60\ BE)}$$

where n = number of errors.

This relationship is shown as curve 1 in Figure 2. From this curve the following can be deduced:

a) If for example 93% of one minute periods has to meet the 1×10^{-6} Mean BER, a long term Mean BER of better than 4×10^{-7} would be required.

b) A long term Mean BER of 1×10^{-6} would yield only 46% of the one minute periods meeting the 1×10^{-6} Mean BER criterion.

If the measurement period To is extended to 5 minutes at 64 kbit/s, 19 or less constitutes the acceptable error count, equivalent to a Mean BER of 1×10^{-6}. The probability of this occurring is shown as curve 2 in Figure 2. Again, if 93% of five minute periods had to meet this criterion a long term Mean BER of $< 7.2 \times 10^{-7}$ would be required.

Similarly, for a measurement period To of 1h, the equivalent values are 230 or less error counts and a long term Mean BER of $< 9 \times 10^{-7}$, to give 93% of 1h periods meeting the 1×10^{-6} Mean BER criterion. This is shown as curve 3 in Figure 2.

Curves 1, 2 and 3 of Figure 2 can be used to relate the percentage of time T_L and various averaging periods To with a Mean BER of 1×10^{-6} to the long term Mean BER. By using Figure 1, the long term Mean BER can then be translated into % EFS.

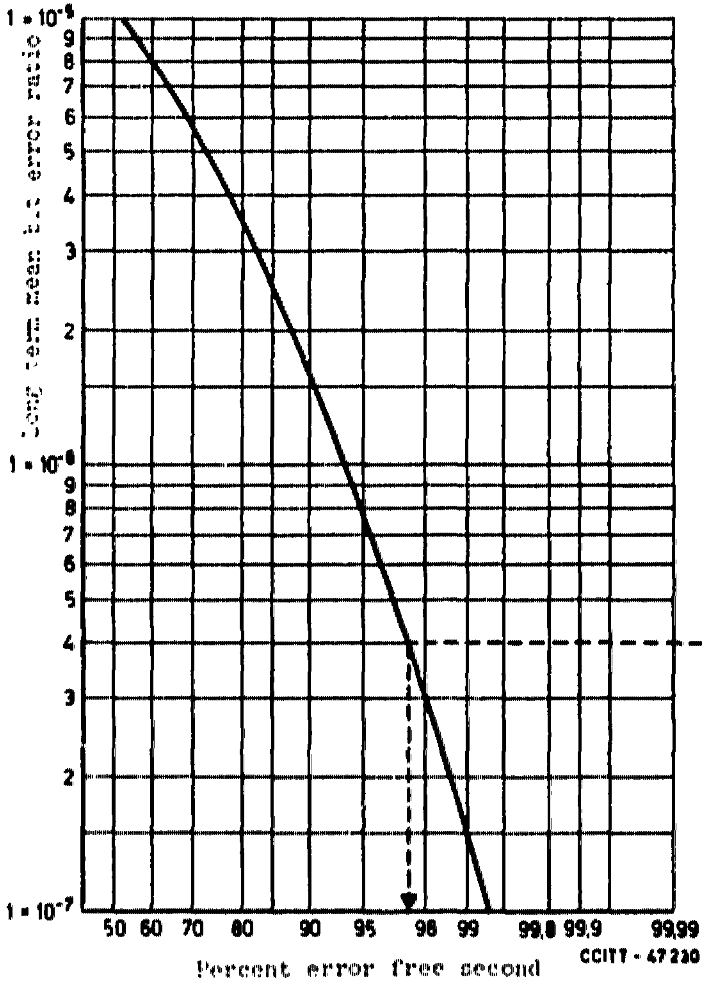


Figure 1 - Relationship between percent error free seconds and long term mean bit error ratio
(Assumption : Bit errors poisson distributed)

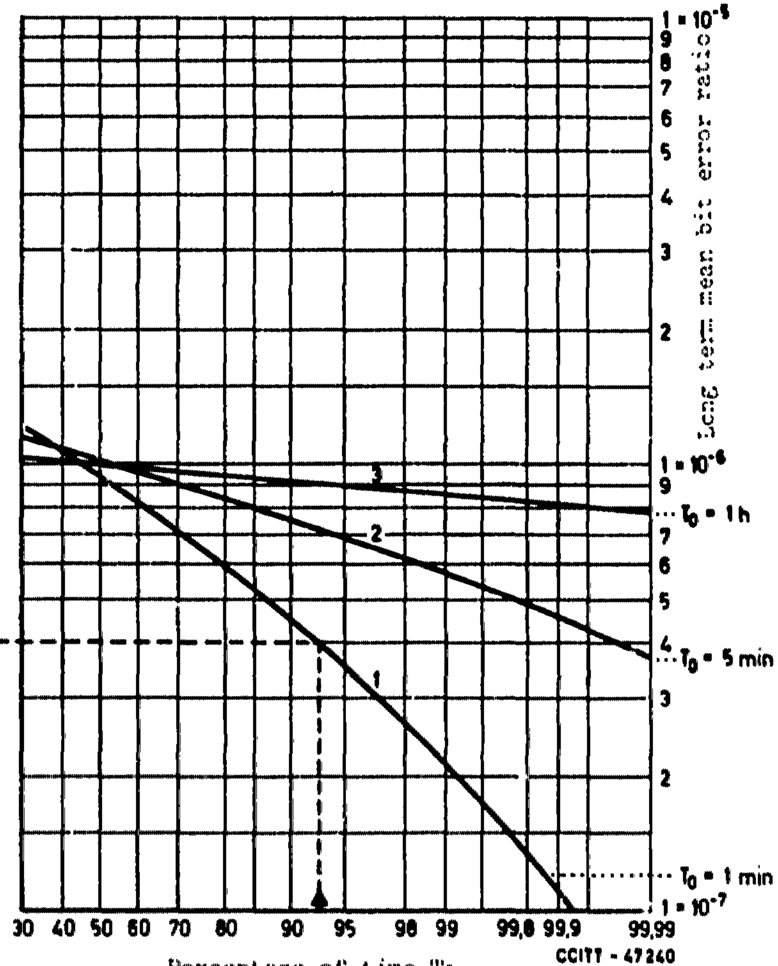


Figure 2 - Relationship between percentage of time T_1 and long term mean bit error ratio for mean bit error ratio of 1×10^{-6} of various averaging periods T_0
(Assumption : Bit errors poisson distributed)

Annex 2

(to Question 9/XVIII)

An approach to the formulation of error performance

(Contribution by the Federal Republic of Germany)

The error performance requirements outlined in the following are based on the idea that data users have a legitimate interest to transmit a large percentage of data blocks without bit errors. However with 64000 bit/s (or 48000 bit/s - the highest data rate specified in Recommendation X.1) these blocks will in general not have a duration of 1 second. In other words, the request to have 95 % error-free data blocks (EFB) is entirely appropriage whereas the request to have 95 % error-free seconds appears to be too stringent.

It may be useful to consider requirements graded in accordance with the length of data blocks (or more generally, of intervals of certain numbers of bits). About five interval lengths, for instance from 200 to 20000 bits, might be suitable to cover the actual conditions of remote data processing.

Assuming Neyman's "Type A contagious distribution", which describes clustered distributions, symbols n = number of bits in the chosen interval, p = long-term mean error ratio, m_1 = cluster - bit ratio (i.e. relative frequency of error events per stated sample size of transmitted bits), and m_2 = mean number of errors per cluster, with $m_1 m_2$ = np, one obtains the probability of an interval containing no errors as

$$p_o = \exp \left[- \frac{np}{m_2} (1-e^{-m_2}) \right]$$

and specifically, with p = 5.10^{-6} (which leads to > 90 % of minutes with a bit error ratio of $\leq 10^{-5}$, i.e. < 38 errors per minute), and m_2 = 2, i.e. a conservative value

p_o = 99.95 % for 200-bit intervals

p_o = 95.76 % for 20000-bit intervals.

Based on the above reasoning, it is proposed to consider, as an alternative to the concept of "error free seconds", the approach of a graded system of requirements based on a set of different block (or interval) lengths. Using this approach, it should be possible to reconcile the requirements of data transmission with a mean bit error ratio of 5.10^{-6} (as sufficient for telephony).

Furthermore, the actual distribution of bit errors (probably characterized by error bursts, or clusters) should be studied before finalizing Recommendation G.821. Since a large percentage of the errors will be allocated to the subscriber lines, results of error measurements on such lines are urgently needed.

Annex 3

(to Question 9/XVIII)

At the instigation of Study Group VII, the concept of error-free seconds has been used in the preparation of Recommendation G.821. It was pointed out by one Administration that future studies in Study Group VII may show that this is not the most appropriate method of defining the performance requirements for data and other non-voice services.

Annex 4

(to Question 9/XVIII)

Jitter accumulation on digital paths and jitter performance of the components of digital paths

(Contribution from Federal Republic of Germany)

Summary

In the present contribution two models of digital paths are considered. Starting from the jitter characteristics of the components of a digital path, the jitter accumulation along a path is described with the aid of these models.

It is shown that for unrestricted interconnection at international interconnection points jitter characteristics are required which are not necessary in national networks.

1. Digital path models for jitter studies

Studies which started from the preliminary considerations of the contribution COM XVIII-No. 69 resulted in two models of digital paths which are particularly suitable for the treatment of the pending jitter problems and applicable to all hierarchical bit rates.

1.1. Digital path model a)

One of the path models - model a) of Fig. 1 - is a tandem connection of line path and muldex sections which alternate systematically.

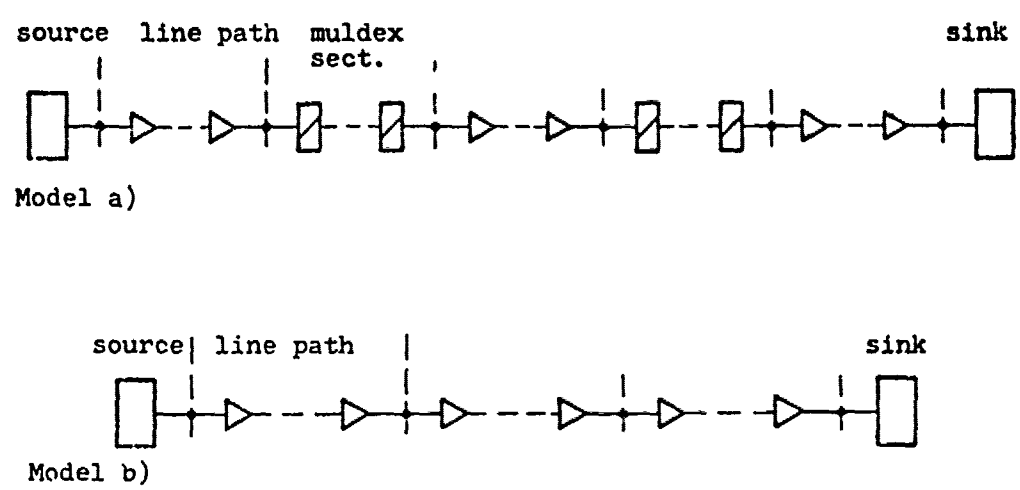


Figure 1 - Digital-path models for jitter studies

The fact that the components of digital paths can be divided into two groups with basically different jitter behaviour has a favourable consequence:

- PCM multiplexers, digital multiplexers, digital demultiplexers, and digital exchanges tolerating relatively high input jitter, but causing low output jitter;
- line paths (i.e. line paths on cables or other media) tolerating low input jitter, generating inherent jitter and, therefore, producing relatively high output jitter.

Model a) is sufficient for the description of the conditions in a future digital network of the Deutsche Bundespost. The same is also expected for many other national networks.

1.2. Digital path model b)

At international interconnection points, it should be possible to interconnect without restrictions sections of the same hierarchical bit rate via the internationally specified interfaces. This means, a direct interconnection of line paths with respect to jitter must be possible as worst case. Therefore model b) of Fig. 1, where the digital path is exclusively built of line paths, represents - in addition to model a) - an adequate model for international interconnection in jitter studies.

2. Jitter accumulation on a digital path

With given output jitter in the absence of input jitter and given jitter transfer function, the shape of jitter along a digital path can be determined.

The lower limit of the maximum tolerable sinusoidal input jitter of a path must be situated with a clearance above the largest expected output jitter of the preceding line path.

In the following, the jitter increase along the path is represented for the path models as per 1. The response has been determined by calculation and by measurements. The results are illustrated in general form, independent of the bit rate.

2.1. Model a)

In Fig. 2, the output jitter of a muldex section and of a line path from model a) is represented without input jitter. Fig. 3 represents the associated jitter-transfer functions.

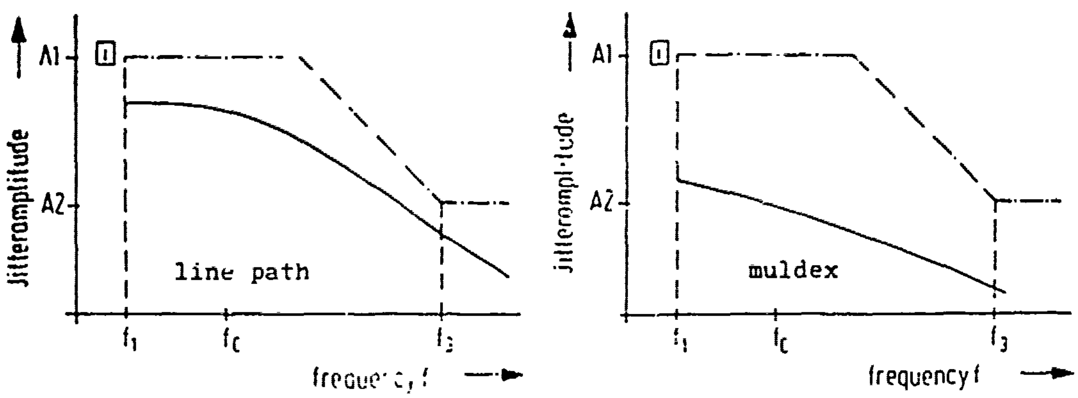


Figure 2 - Output jitter of a line path and a muldex section

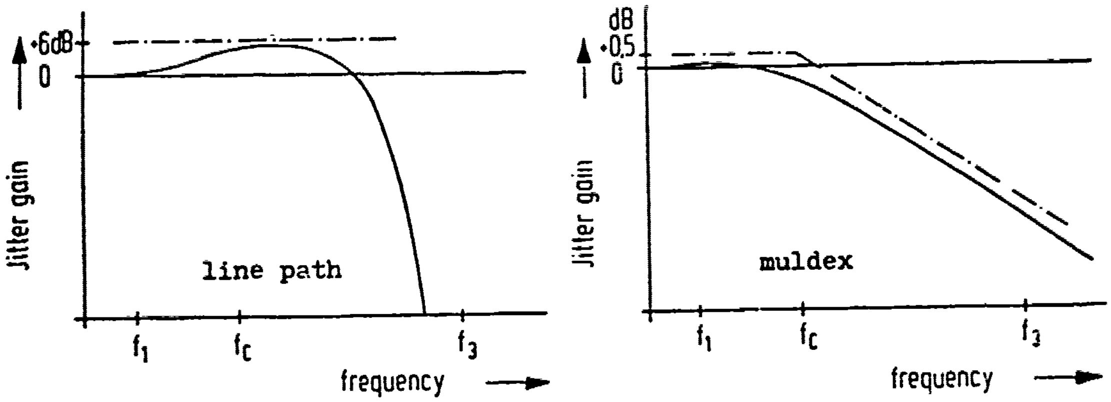


Figure 3 - Jitter-transfer function of a line path and a muldex section

Figure 4 shows the amplitude of phase jitter along a path in accordance with model a)

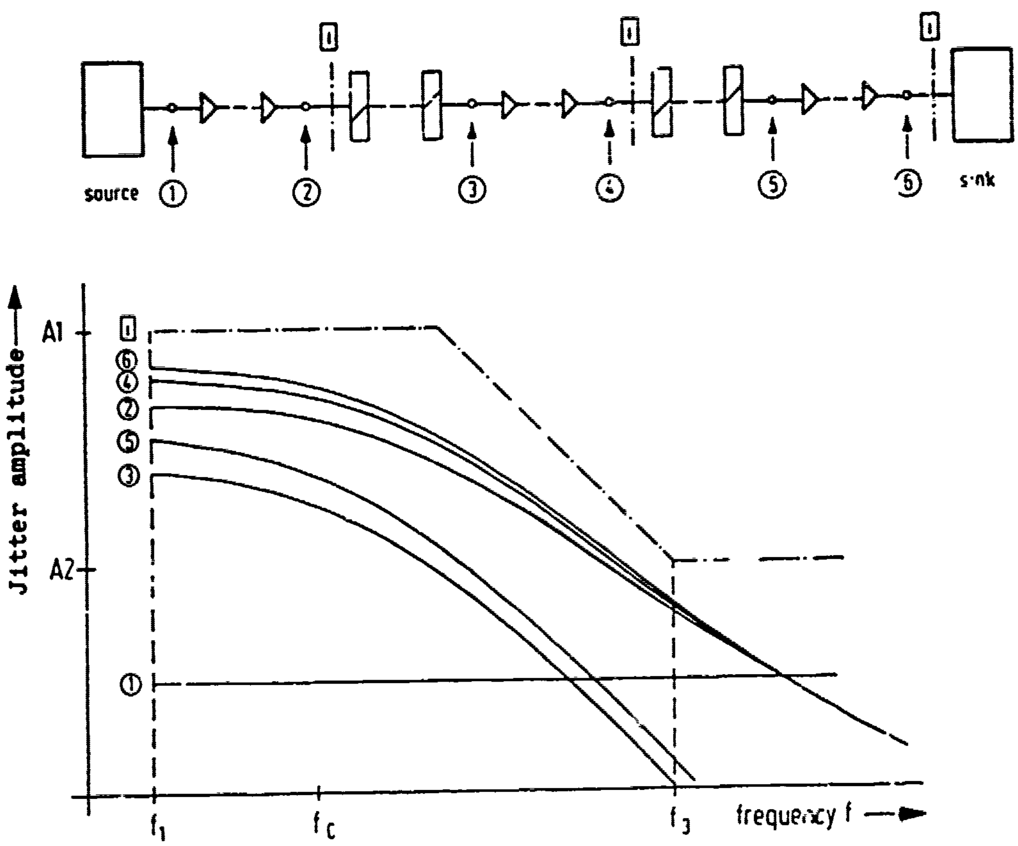


Figure 4 - Output jitter in model a)

Source:	Pseudo-random pattern
Output jitter:	Measured via high-pass filter with variable cut-off frequency f
1	Input jitter according to Rec. G. 703 (shown for comparison)
1	Output jitter of the source
2	Output jitter of the first line path, corresponding to the output jitter of a line path without input jitter
3	Reduction of the input jitter due to the first muldex section
4	Output jitter of the second line path. (Due to jitter accumulation below cut off frequency f_c, the values are slightly higher than the ones obtained for line path without input jitter
5	Reduction of input jitter due to the second muldex section
6	Output jitter of the third line path

In the case of paths with more sections than in model a), the jitter values would increase, although slightly.

2.2. Model b)

In model b) three line paths are connected in tandem. Fig. 5 shows that, for the same line path as in model a), the output jitter amplitude will rise as in Fig. 4. It can be seen that the output jitter amplitude of the last line path in transmitting direction increases permanently with the number of line paths ahead.

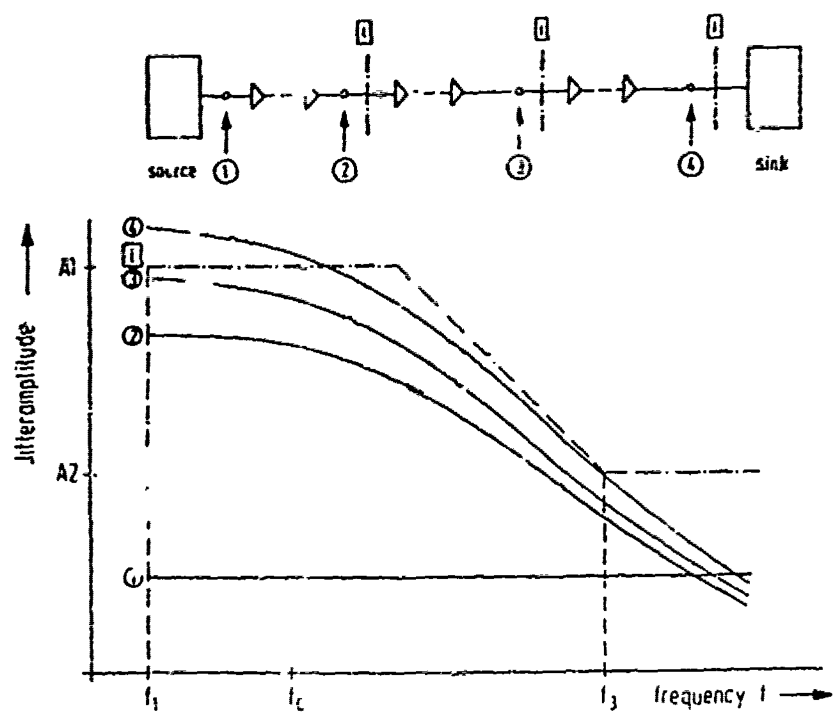


Figure 5 - Output jitter in model b) without jitter reducing means

[1] Input jitter according to Rec. G.703 (shown for comparison)

① Output jitter of the source

② Output jitter of the first line path (identical ② in Fig. 4)

③ Output jitter of the second line path

④ Output jitter of the third line path

In order to control jitter accumulation over the entire path according to model b), jitter reducing means are necessary within the second and within all following sections of the path. The reducing means can be considered a replacement for the jitter reducing effect of the demultiplexer in model a). Consequently, the jitter-transfer function (Fig. 6) describes the muldex - line path - cascade. It is the product of the jitter transfer function of the components muldex and line path (Fig. 3) from model a).

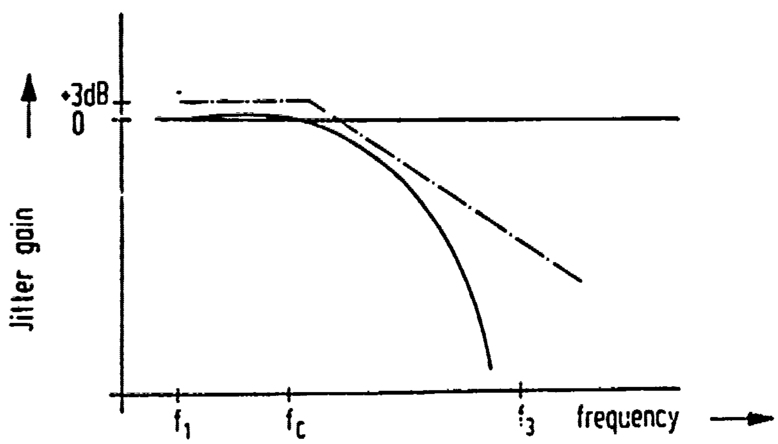


Figure 6 - Jitter transfer function of a line path in model b)

Additional means for jitter reduction are not necessary if the output jitter of a line path without input jitter is already substantially below the maximum tolerable value.

The necessity of specifying the jitter transfer function of a line path is the consequence of unrestricted interconnection at international interconnection points.

Fig. 7 shows the shape of the jitter amplitude along a path according to model b).

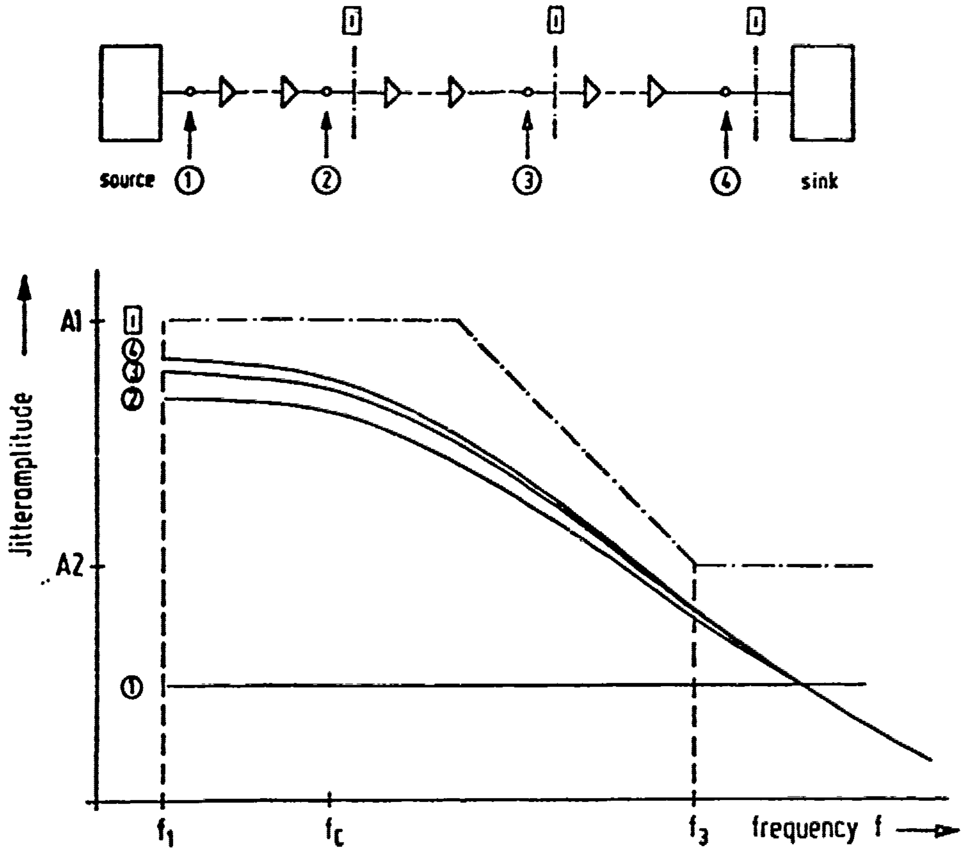


Figure 7 - Output jitter in model b)

Source:	Pseudo-random pattern
Output jitter:	Measured via high pass filter with variable cut-off frequency f
i	Input jitter according to Rec. G.703 (shown for comparison)
①	Output jitter of the source
②	Output jitter of the first line path (identical ② in Fig. 4)
③	Output jitter of the second line path
④	Output jitter of the third line path.

In paths with more than three line paths, the jitter amplitude increases at the end of the path but slightly because of the jitter-reducing effect of the maximum tolerable jitter transfer function of the line path.

3. Jitter specifications

It has been shown in chapter 2 that the components of a digital path are completely characterized by

- tolerable sinusoidal input jitter
  (more exactly: lower limit of maximum tolerable input jitter)

- maximum output jitter in the absence of input jitter, measured as peak-to-peak value with a high-pass filter
- jitter transfer function, measured as jitter gain with sinusoidal input jitter.

In Fig. 2 to 7 the boundaries for jitter behaviour (dash dotted lines) are drawn in accordance with calculations and measurements of tolerable jitter responses.

Annex 5

(to Question 9/XVIII)

State of CCITT studies regarding equipment design objectives, in respect to error performance, for digital transmission systems on cables

1 Introduction

Equipment design objectives for digital transmission systems on cables (balanced pair, coaxial pair and optical fibre) are specified in Rec G.911-918 and G.921-922, alongside network performance objectives for digital line sections at the corresponding bit rates.

CCITT studies have not yet advanced to the stage that the clauses of those recommendations which are to relate to error performance can be completed.

The purpose of this annex is to present the current status of those studies so as to provide designers of cable digital transmission systems with interim guidance pending the completion, in respect of error performance, of the mentioned Recommendations.

2 Relationship to network performance objectives

The network error performance objectives for the integrated services digital network (ISDN) are stated in Rec. G.821 (Error performance of an international digital connection forming part of an integrated services digital network). Those objectives take account of all errors, however caused, liable to occur in a digital network. Equipment design objectives for digital transmission have to be compatible with the network objectives of Rec. G.821

Most errors in a digital network are attributable to interfering influences such as lightning strikes, induction from electricity supply or traction systems or parallel transmission systems, man-made disturbances, etc. Equipment design objectives for digital transmission systems should ensure a degree of provision in the system design to minimise the effects of such interferences. This is a subject of study within new Q. 12/XV. Until such time as the important error-causing interfering factors can be taken into account in the definition of the environmental conditions forming part of the equipment design objectives system designers need to allow generous margins in respect of them. This is done within both of the design approaches described hereafter.

However, it should be observed that radio relay systems and satellite systems will be subject to different error characteristics than those incurred by metallic cable systems.

3 Available equipment design objective approaches

Two distinct approaches to the error performance aspects of digital transmission system design have been identified:-

- The error performance of a hypothetical reference digital path of defined length and constitution is specified; this serves as a guide to the design of all equipment items involved in the hypothetical reference digital path. This approach is analogous with that adopted for specifying the noise performance of analogue transmission systems.

- The quality of repeaters used in a transmission system is specified and measured in terms of "repeater margin".

Both approaches involve the definition of environmental conditions. In either case it is impracticable, for the present, to include the environmental factors considered in 2 above.

Although based on different philosophies it is not to be assumed that either approach necessarily excludes the other. The final outcome of CCITT studies may well be clauses on error performance equipment design objectives in which the two approaches complement each other.

The two approaches are considered separately and in detail in Sections 4 and 5 respectively.

4 Definition of error performance equipment design objective, specified in relation to a hypothetical reference digital path

4.1 Rec. G.821 expresses the overall objective for the error performance of the digital network, ie a 25000 km hypothetical reference digital path of 64 kbit/s in terms of the percentages of:

- error free seconds (data requirement)

- minutes having a bit error ratio better than a threshold value (the telephony requirement).

Clearly, equipment design objectives for digital line systems have to be consistent with these overall network objectives; two important adaptations are considered necessary.

Firstly, the equipment design objective for digital line systems is best expressed as a "long term mean bit error ratio". System designers invariably start with an objective for this. They then design regenerators etc taking account of the known impairments such as the diminishing of eye openings, noise, crosstalk, level inaccuracies etc.

Secondly, there needs to be a margin for the errors due to unknown hazards outside the control of the designer, eg unquantifiable interferences from external sources, so that the overall result is acceptable.

4.2 Annex 6 to Q. 9/XVIII (Abridged version of ITT contribution COM XVIII - No 369) suggests that network objectives expressed as in Recommendation G.821 can, with acceptable accuracy, be alternatively expressed in various combinations of:-

- a long term mean bit error ratio

- a clustering index (ie mean errors per cluster).

It seems not possible, as yet, to make a firm judgement as to the degree of clustering which will be exhibited by digital networks. At the same time, several contributions have indicated that it is safe to assume, for networks based on cable systems, at least a modest level of clustering. For a long term mean error ratio of 5.10^{-6} the percentage of error free seconds can be expected to exceed 90% for values of clustering index greater than 3 and the percentage of minutes of error ratio better than 10^{-5}* to exceed 90% for values of clustering index less than 10.

A long term mean error ratio of 5.10^{-6} would seem to be a reasonable assumption for the total errors, over 25000 km, to which equipment design objectives might be related. This value is not necessarily appropriate for digital systems other than those using cable.

4.3 A good proportion of the errors in a 25000 km 64 bit/s connection will be the results of external interfering factors, protection against which is best provided by such measures as screening of the cable and repeaters, high low-frequency cut off in the repeater amplifier, etc. It is a matter for future study to establish, within the scope of the equipment design objectives, appropriate standards for such aspects of digital transmission system design.

Pending the outcome of such studies, it is suggested to allow for external interferences by means of a generous margin between the design objective specified for the error performance of a hypothetical reference digital path, functioning in an interference-free environment, and the proportional allocation to that path of the network performance objective for the 25000 km hypothetical reference connection. As a provisional solution, subject to revision in the light of further results of error measurement programmes 80% of the total acceptable errors might be appropriated for such margin.

This leads to a long term mean bit-error ratio of 1.10^{-6} as an appropriate objective for a 25000 km hypothetical reference connection composed, in all its links, of digital line systems on cable and operating free of external interferences.

The following table shows how this bit error ratio might be allocated to the main sub-divisions of the total hypothetical reference connection.

---

* The value presently proposed in Rec. G.821 is 1.10^{-6}.

Allocation of design error rate for a maximal length hypothetical reference connection

Item	Quantity	Allowance (in 10^{-8})
International homogenous high rate line link, eg - coaxial cable (land or ocean) - optical fibre	25000 km	25
National links: - high rate - primary centre to subscriber	1500 km (750 km each end) Not specified	33 each end Total 66
Digital exchanges: - international - national and multiplexing equipment	7 4 each end	Nominally zero
		Total 91

The objective error performance for those hypothetical reference digital paths of Rec. G.721 appropriate for digital line systems providing international high rate line links might then be derived as follows:-

Digital line system			Hypothetical reference digital path		Suggested error ratio design objective (exc. allowance for multiplexers)
Rec	Capacity (Mbit/s)	Medium	Rec G. 721	Length (km)	
G.916 II	34 368	Coax. pair	Fig 3	2500	$2.5 \cdot 10^{-8}$
G.918 II	139 264	Coax. pair	Fig 4	2500	$2.5 \cdot 10^{-8}$

Objectives for systems providing national links may be derived similarly, taking into account the global allocation and the constitution of the relevant HRDP.

5 Definition of error performance equipment design objective based on repeater margin

5.1 General observations

In order to specify and evaluate repeaters, multiplexes, and actual or hypothetical combinations of these (as, for example, in a hypothetical reference circuit) it is necessary to adopt measures of performance. For analog equipment, noise power has been the principal measure employed. In the digital realm, there has been a tendency to use error ratio for this purpose

Error ratio is far from an exact analogy to noise power, however. For example note that an analog repeater or multiplex typically injects a fairly constant noise power, which is close to the design value. A digital repeater or multiplex, however, typically operates at zero error ratio, and when it makes an error these are usually at a much higher rate than whatever design rate may have been specified, and are indeed often a result of disturbances not related to the basic system parameters. As a consequence, error ratio has several drawbacks as a measure. Some of these are:-

- System components almost never operate near their design error ratios.
- Errorratio of a multiplex is an almost meaningless concept. Except in case of failure, multiplexes should operate error free.
- Errorratio is a very poor measure of repeater performance. Observed errors in a properly designed and operated repeater are most likely a result of electromagnetic or other disturbances unrelated to the repeater.
- The allocation of errorratios to the several segments of a complete connection bears little relation to situations observed in practice. Typically at any time when the errorratio of a real connection is significant, the errorratios on all segments except one are negligible.

It is interesting to note that noise power as a measure of performance in analog systems shares none of those drawbacks.

As a consequence of these considerations G.821 expre. es the objective for a hypothetical digital connection in terms not of error rate, but in % of interval (one second and one minute) which have errors in excess of a stated number.

5.2 Repeater margin

In analog systems, the same measure, noise power, used in evaluating connections is also a good measure of repeater quality and is often used as an objective. In digital systems neither error ratio, nor either of the measures of G.821 bear much relation to repeater quality. The measure suggested for this purpose is margin (expressed in dB) against a 10^{-7} error probability. The margin that increase in the dominant interference (such as thermal noise or crosstalk) from its nominal or actual value which produces a 10^{-7} error probability. This of course requires careful definition and some contributions have addressed this issue. Regardless of exact definition, however, the crucial point is that the repeater is specified and measured under conditions excluding any uncontrolled interference (such as lightning) so that repeater error statistics are in fact well represented by Poisson arrivals and so error probability is complete description of the error performance. This is in sharp contrast with a typical repeater installed in the field.

The choice of 10^{-7} is not crucial - but is probably quite reasonable for higher speed systems. 10^{-6} might be more appropriate for repeaters in the 1.5-2 Mbit/s range, so that measurement intervals of the order of a few seconds can be used.

To illustrate the utility of this measure, consider a 100 Mbit/s coax repeater with margin of 5 dB against 10^{-7} error probability.

Given a repeater in the lab or in the field, there is no difficulty in measuring its performance (by adding noise for example) as the added noise can readily be adjusted until 10^{-7} (at 100 Mbit/s $\simeq$ 10 errors/second) is obtained. (Actually the measurement of error probability is not critical, as an order of magnitude variation corresponds to only about 0.5 dB of margin). The margin against 10^{-7} error probability is also a reasonable criterion for a designer to consider, and a convenient quantity to use in production testing.

It might be thought that the repeater could be specified by its experienced error probability (with no additional noise) - but this would be about 10^{-18}, so low as to be completely unmeasurable.

As a tentative design objective, a repeater margin of the order of 5 dB against 1.10^{-7} to 10 dB against 1.10^{-6} might indeed be reasonable for repeaters above 3 Mbit/s. In the 1-3 Mbit/s range, this could be against 10^{-6} error probability.

These values are probably not grossly unreasonable but they are not directly obtained from the network performance objectives. The relationship should be the subject of further study.

6 Commissioning objectives

6.1 Checks of system error performance

Both the approaches above relate to performance in an interference-free environment. It follows that when the performance of an installed system is measured with a view to checking whether a design objective is met precautions are necessary to identify errors attributable to the environment.

This is necessarily less than an exact procedure, and primary reliance should be placed on verification that the components meet their design objectives.

Periods during which errors occur simultaneously with observed potential interferences should clearly be discounted.

Consideration of the distribution of measured errors may justify the discounting of isolated bursts, even when it has not been possible positively to identify causes.

Equipment design objectives are applicable only to systems in good working order. It follows that errors caused by faults have to be discounted; however, the overall incidence of interruptions due to faults is properly subject to recommendations concerned with equipment reliability.

Some errors may result from components failing intermittently or undergoing changes with respect to their operating characteristics. Such might justifiably be attributed to equipment design rather than reliability failure.

A satisfactory text on commissioning objectives for error performance taking all these aspects into account, can be drafted only after further study and experience.

Annex 6

(to Question 9/XVIII)

Error performance objectives for integrated services digital networks (ISDN)

(Contribution by the International Telephone and Telegraph Corporation)

1. Introduction

The digital connections set up by an integrated services digital network, normally of 64 kbit/s, are envisaged to be used for different purposes, speech, data, facsimile, etc. These usages differ in their sensitivities to errors introduced into connections by the network, both as regards the numbers of errors and their distribution. For this reason the error performance requirements of the ISDN are being formulated in terms of mean error ratios and integrating periods which are particular to each usage. For data the objective is expressed in error-free seconds; for telephony by means of minutes having better than a threshold error ratio. Other envisaged ISDN usages, facsimile, picture phone, etc. may require the formulation of requirements expressed in yet other ways.

The 64 kbit/s connections of the ISDN should, of course, have an error performance which is at once acceptable for all ISDN services. To this end it is desirable that the separately expressed objectives be compatible in that they conform to a single required quality of transmission.

The objective of this contribution is to present a means of defining error performance which at once

- takes account of the ways errors are distributed in practice
- is conveniently convertible into forms which are meaningful for different ISDN usages
- provides a basis of comparison of the levels of digital transmission quality implied by usage-oriented performance objectives.

2. Assumptions relevant to error distribution

Early studies of digital network error performance generally assumed a constant bit error probability, its value being approximated by the long term mean error ratio, a readily measurable quantity. The mathematics of the resultant error distributions was available - the Poisson distribution. The percentages of integrating periods, seconds, minutes, 5-minutes or whatever, during which specified error ratios, choices exercised according to the particular ISDN service being considered, might be exceeded were calculable from the long term mean error ratio.

In the event measurements of realized error performance have demonstrated that errors, in practice, occur in clusters, so invalidating the assumption of a constant bit error probability and derivations based on Poisson theory.

We suggest that the following alternative basic assumptions accord reasonably well with the actual behaviour of real networks, so far as they have been observed ;

- The probability per bit that an error cluster will begin is constant and is approximated by the long term mean ratio of cluster rate to bit rate
- The numbers of errors in clusters themselves follow a Poisson distribution.

The mathematics of such error distributions, although rather more complex than the simple Poisson distribution, is already available and promises to satisfy the objectives put forward in the introduction above.

3. Neyman's Type A Contagious Distribution (see ref.)

Such a distribution, derived from the foregoing stated assumptions, is defined by two parameters:

- the cluster-bit ratio (m_1) expressed as the mean number of clusters per stated sample size of transmitted bits
- the mean number of errors per cluster (m_2).

The probability of encountering exactly "r" errors in a sample is given by:

$$p_r = \frac{m_2^r}{r!} \cdot e^{-m_1} \left\{ \sum_{i=0}^{\infty} z^i \frac{i^r}{i!} \right\}$$

where $z = m_1 \cdot e^{-m_2}$

The special case of the probability of a sample containing no errors is given by:

$$p_0 = \exp\left\{-m_1(1 - e^{-m_2})\right\},$$

The mean of the distribution is $m_1.m_2$; this is equal to n.p, where n is the number of bits in the chosen sample, 64 kbits possibly, and p the long-term mean error ratio.

The variance of the distribution is $m_1 m_2 (1 + m_2)$.

It is to be noted that as $m_2 \rightarrow 0$ so do the mean and the variance of the distribution approach equality, the characteristic of the simple Poisson distribution.

Also noteworthy is the fact that clusters are characterized only by the numbers of errors they contain,

For 64 kbit/s transmission, long term mean error ratios of 10^{-5}, 5.10^{-6}, 3.10^{-6} and 1.10^{-6} and mean values of errors per cluster of from 1-1000 we have calculated the forecast percentages, according to the foregoing mathematics, of:

- error free seconds (Fig. 1)
- minutes during which the error ratio is better than 10^{-5} (Fig. 2)
- minutes during which the error ratio is better than 10^{-6} (Fig. 3).

The ranges of values taken into account are those appropriate to be considered in relation to performance objectives for a 25,000 km hypothetical reference digital connection, the subject of Rec. G.821 (Error performance on an international digital connection).

percentage
of error-free
seconds

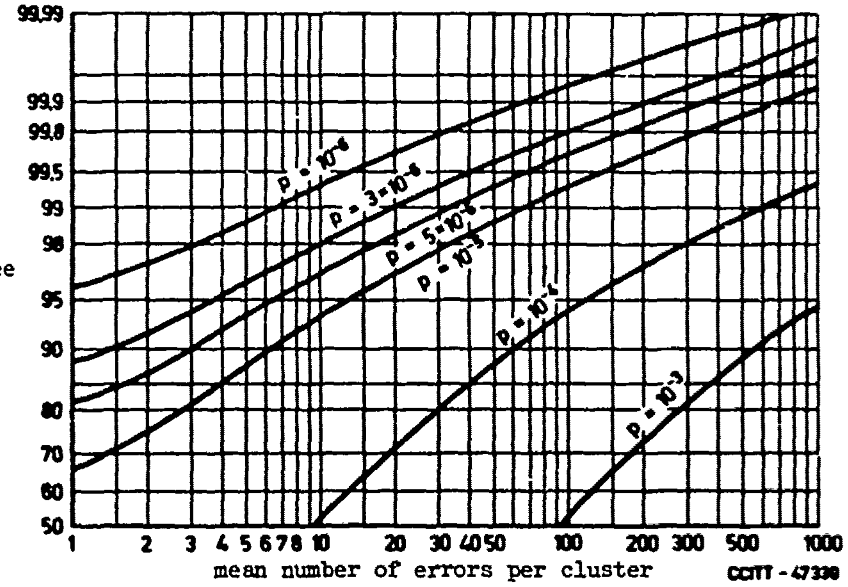


Figure 1 - The variation with degree of clustering of the percentage of seconds of 64 kbit/s transmission which are error-free for different values of long term mean error ratio (p)

percentage
of minutes
of error
ratio < 10^{-5}

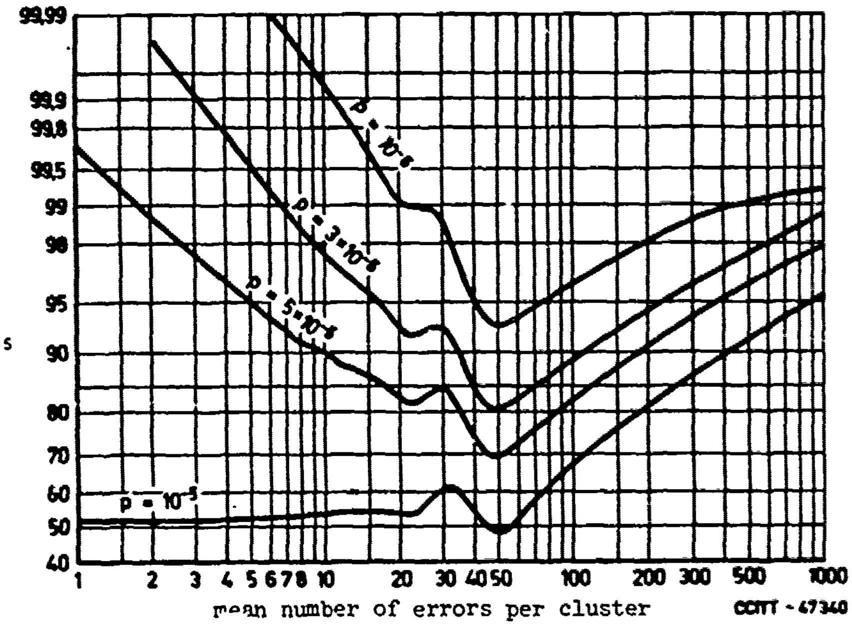


Figure 2 - The variation with degree of clustering of the percentage of minutes of 64 kbit/s transmission having an error ratio less than 10^{-5} for different values of long-term mean error ratio (p)

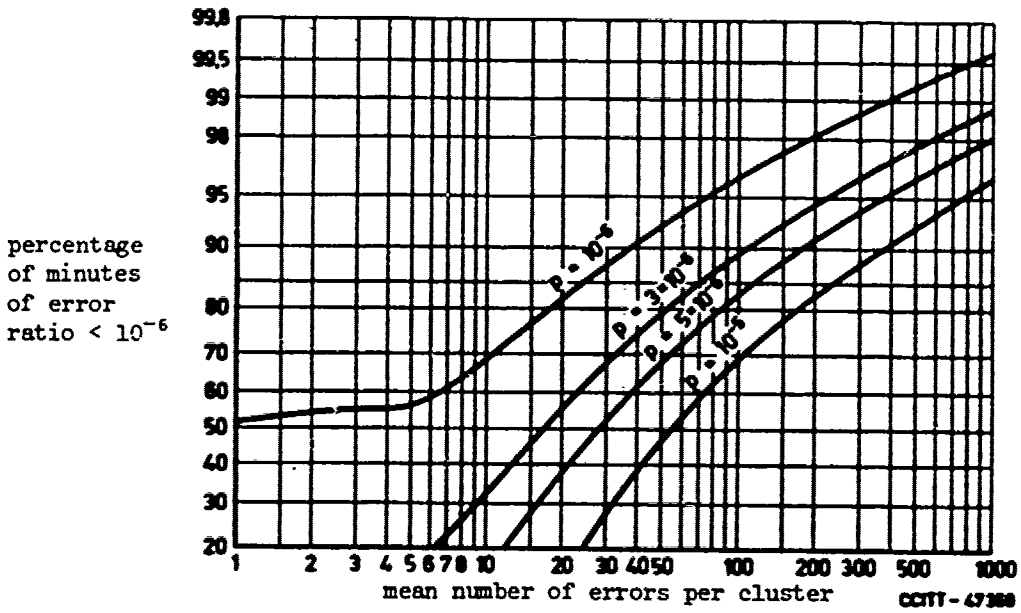


Figure 3 - The variation with degree of clustering of the percentage of minutes of 64 kbit/s transmission having an error ratio less than 10^{-6} for different values of long term mean error ratio (p)

4. Error performance objective for data

Figure 1 confirms the frequently observed effect of error clustering that performance expressed in terms of error free seconds is significantly better than would be expected from a measured long term mean error ratio and an assumed Poisson distribution of errors.

The particular long term mean error ratio which would satisfy a 95% error free seconds objective depends on the degree of clustering. A long term error ratio of 10^{-5} would suffice given a degree of clustering as high as is represented by an errors per cluster value of 12.5.

Otherwise, the necessity of a superior long term mean error rate is indicated.

5. Error performance objective for telephony

Figure 2 demonstrates an effect of error clustering which seems not to have been so well anticipated as that referred to in 3. above but is quite explicable, nevertheless. If the significant integrating duration is one-minute and the acceptability criterion is an error ratio, over the minute, of 10^{-5}, error clustering up to a mean value of errors per cluster of about 50 actually reduces the percentage of acceptable minutes for a given long term mean error ratio. For values of mean errors per cluster greater than 50 the percentage of acceptable minutes increases progressively.

The same effect is not so evident for a one-minute threshold of error ratio 10^{-6} (see Fig. 3). The maximum acceptable number of errors per minute in this case is only three so the curves have shapes rather similar to those for error free seconds (Fig. 1).

6. Need for field measurements of error ratios and distribution:

A major difficulty associated with the formulation, at the present time, of error performance objectives for the ISDN is the scarcity of measurements of the error performance of working systems. Such data are essential if the objectives set are to be realistic.

During the 1977-80 study period to date the only measurements contributed to Study Group XVIII, to our knowledge, were those by France (COM XVIII - Nos. 28 and 278) and Switzerland (COM XVIII - No. 208); these were necessarily limited in scope to relatively short distance transmission systems but, even so, have been extremely useful. It is desirable that the reply to Q. 1/XVIII for 1977-80 and the texts of new questions set for 1981-84 emphasize the importance of new information derived from field measurements.

This contribution confirms the need to measure data which are indicative of the distribution, or clustering, of errors as well as the overall ratio of errors to bits transmitted. This might be done in various ways.

The Swiss contribution COM XVIII-No. 208 analyses the total duration of a 2048 kbit/s transmission into the elemental periods, seconds or minutes, containing different numbers of errors, including zero. The transmission rate and the length and type of connection have to be stated, of course. The French contribution COM XVIII-No. 278 describes a rather similar programme of measurements performed on 140 Mbit/s systems. Both parameters m_1 and m_2 can be determined from the data produced by such measurements.

Alternatively, the parameter m_2 may be measured directly by a counting of the total bits in error and of the number of clusters, i.e. the number of errors which are the first errors of new clusters. A cluster is characterized by the large numbers of correctly transmitted bits which separate it from adjacent clusters compared with the numbers which separate the bit errors within itself.

7. Summary and Conclusions

The study of ISDN performance objectives, e.g. with reference to a hypothetical 25000 km 64 kbit/s connection, is substantially simplified by two basic assumptions:

- that error clusters (as distinct from the errors themselves) are randomly distributed,

- that the numbers of errors per cluster are also randomly distributed.

Error performance objectives expressed in terms of different error ratio thresholds and integrating periods, meaningful for different ISDN services, are then comparable on the basis of two parameters, the long-term mean error ratio and the mean number of errors per cluster. The recommending of inconsistent objectives for different services can in this way be avoided.

The two mentioned parameters do, in themselves, provide a sufficient method of specifying an ISDN performance objective;

furthermore, an objective expressed in this way is readily apportioned, as appropriate, to diffe.ent parts of the reference connection.

The importance of more field measurements of error performance is emphasized; suggestions are made as to how these may most effectively provide information on error distributions as they occur in reality.

REFERENCE

Neyman, J. "On a new class of 'contagious' distributions applicable in entomology and bacteriology". Ann. Math. Statist., **10**, 35 (1939).

Annex 7

(to Question 9/XVIII)

Relation between error measures

(Contribution by American Telephone and Telegraph Company)

1. A proposed draft recommendation suggests two error performance objectives, both of which are to be met concurrently. These are:

- More than 90 percent of minutes to have better than $1 \cdot 10^{-6}$ error ratio.
- More than 92 percent of seconds to be error free (for 64 Kb/s connections).

It is natural to inquire as to the relation between these requirements. This relation may be investigated by making some assumptions regarding the error statistics. Alternatively, the limits on one measure guaranteed by the other regardless of the statistics may be calculated, and this calculation is carried out below. It is found that a connection providing exactly 90 percent minutes at better than $1 \cdot 10^{-6}$ error ratio will provide from 85.5 to 99.8 percent error-free-seconds at 64 Kb/s, and that a connection providing 92 percent error-free-seconds at 64 Kb/s will provide from 0 to 97 percent minutes better than $1 \cdot 10^{-6}$.

2. If we have 90 percent minutes better than $1 \cdot 10^{-6}$, then for 90 percent of the minutes there must be less than (10^{-6}) (64) (10^3) (60) = 3.8 errors. Therefore, there can be from 0 to 3 errored seconds in the 90 percent interval. In each minute of the 10 percent (at the limit) of minutes worse than 10^{-6} there must be at least four errors, which could all occur in one second, or there could be 60 errored seconds in each such minute. Therefore, the percent error seconds corresponding to the 90 percent minutes criteria ranges from

$$(0)\ 90\% + (1/60) 10\% = 0.17\% \quad \text{to}$$

$$(3/60)\ 90\% + (60/60)\ 10\% = 14.5\%$$

3. If we have 92 percent error free seconds, the 8 percent errored seconds could each have one or more errors. Since any second with 4 or more errors causes the minute in which it appears to be worse than the 10^{-6} threshold, and 8 percent of 60 seconds is 4.8 seconds, there may be no minutes better than $1 \cdot 10^{-6}$. On the other hand, if three errored seconds occur in each minute for 31 minutes, and then one minute with 60 errored seconds occurs we have still (approximately) 8 percent errored seconds, but now (approximately) 97 percent minutes which are better than $1 \cdot 10^{-6}$.

Annex 8

(to Question 9/XVIII)

Effect of digital signal processing devices on the transmission performance of international digital connections

(Contribution by Bell-Northern Research/CTCA)

1. Introduction

It has been known for some time that an international digital connection between countries using A-law and Mu*-law PCM coding will require digital code conversion if an excessive amount of distortion is to be avoided. A suitable code conversion has been recommended (Recommendation G.711) and the performance of an ideal connection involving an A/Mu, Mu/A-law conversion has been published (CCITT Green Book Vol. III-3 P. 844 Figure 4).

Nevertheless, a practical international connection may contain additional signal processing devices which can cause further distortion. There will also be a further degradation due to random bit errors.

Using computer models, calculations have therefore been made of the overall signal to distortion ratio (for a gaussian input signal) of various practical connections. These connections include A/Mu-law conversions and other digital signal processing devices such as a 6 dB digital pad and a typical bit reduction scheme. The effect of random bit errors has also been investigated. The models assume ideal A and Mu-law segmented coding laws and conversion rules according to Recommendation G.711.

Calculations of the signal to distortion ratio of A and Mu-law encoding with each of the digital signal processing devices alone are given in Figures 1a and 1b. Notice that a 6 dB pad has been used in these calculations. Other pad values would give somewhat different values of signal to distortion ratio for A or Mu-law.

In this and subsequent figures the dashed line represents the template for the allowable signal to distortion ratio of a single codec from Recommendation G.712 (gaussian signal input).

2. Calculation Results

(1) One very important practical connection involves an A/Mu-law conversion where the transmission loss is achieved by means of a 6 dB digital pad inserted prior to the receive codec. This connection is shown in Figure 2a. Signal to distortion ratios for Mu to A and A to Mu directions of transmission for this configuration are shown in Figures 3a and 3b, with and without a 6 dB digital pad and a random bit error ratio of 10^{-5}. It is clear that the degradation introduced by the random bit errors is the dominant feature. In comparison with Figures 4a and 4b, which show the effect of random bit errors without any pad or A/Mu law conversion, it can be seen that the connection of Figure 2a would have the same signal to distortion ratio as a random BER of between 10^{-4} and 10^{-5}. This level of signal to distortion ratio can result in perceived degradation (CCIR Doc. 4/75; 1974-1978), particularly for low level signals.

---

* To facilitate the preparation of this text the symbol Mu was used in place of Greek letter μ throughout the text.

(ii) The configuration analysed above can be extended to include the effect of extra signal processing which might occur due to for example, digital speech interpolation systems. When such a system is in overload one technique to make more channels instantaneously available is to reduce the number of bits/sample to 7. This is also known as trans-coding. Although this bit reduction occurs dynamically depending on the instantaneous talker activity, a worst case is to assume continuous bit reduction to 7 bits. A connection consisting, therefore, of a Mu/A-law conversion, 6dB digital pad, 7 bit transcoding and a random BER of 10^{-5} has been analysed. This connection is shown in Figure 2b. The signal to distortion ratio for either direction of transmission of this connection is also shown in Figures 3a and 3b. The additional 7-bit transcoding causes some reduction in the signal to distortion ratio for high level signals. However, in this level range the signal to distortion ratio is still sufficiently high that the subjective effect would be unnoticeable. Once again, the subjective effect would be on low level signals.

3. Summary and Conclusions

Several computer models of practical international digital connections involving A/Mu law conversions, digital pads and bit reduction schemes have been analysed. Calculations have shown that the addition of digital signal processing devices result in a decrease of signal to distortion ratio. For connections with a bit error ratio of 10^{-5} this decrease is sufficient to cause a noticeable change in the perceived degradation, particularly at low signal levels. It is therefore suggested that the effect of digital processing devices should be taken into consideration when allocating impairment in an international digital connection.

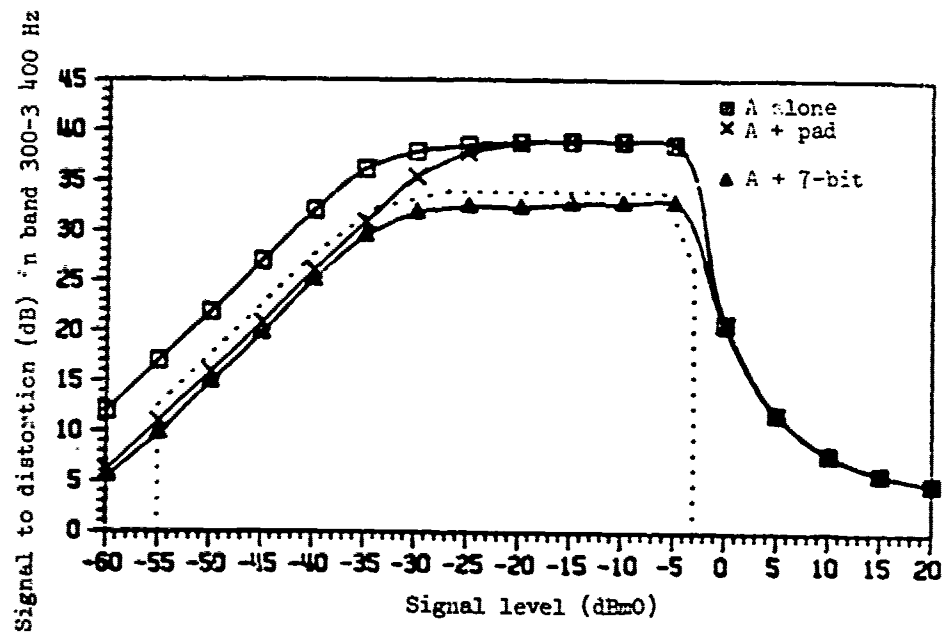


Figure 1a - Effect of individual devices on ideal 8-bit A-law PCM

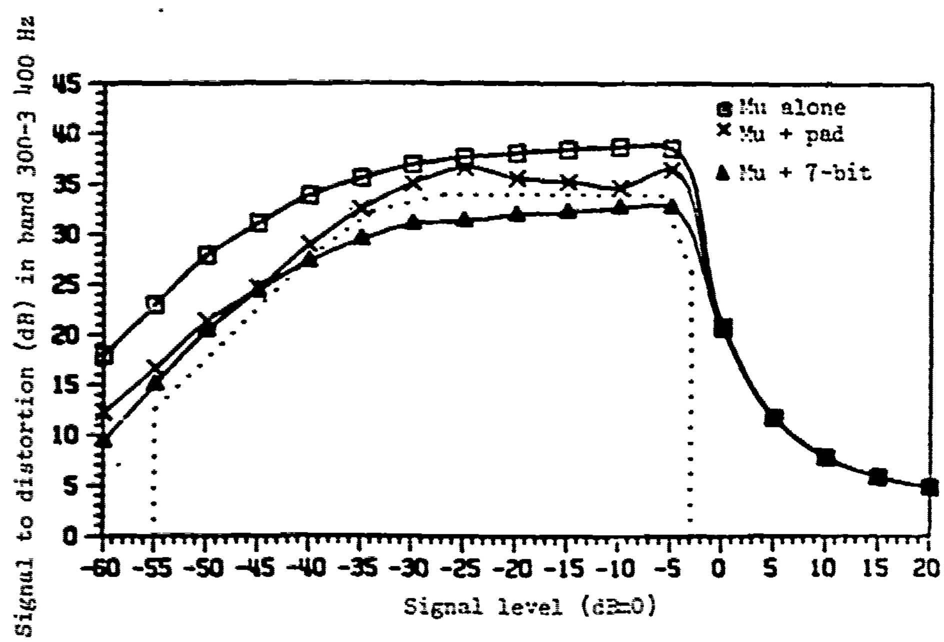


Figure 1b - Effect of individual devices on ideal 8-bit Mu-law PCM

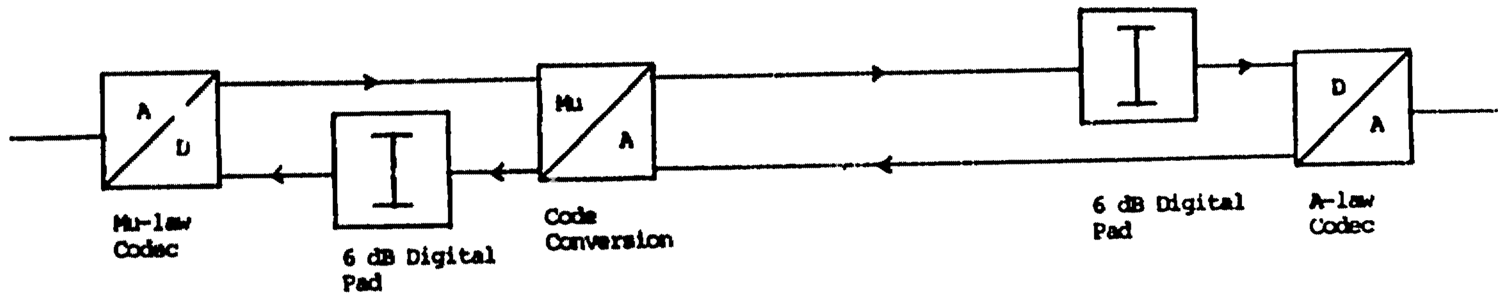


Figure 2a - International connection with 6dB digital pad

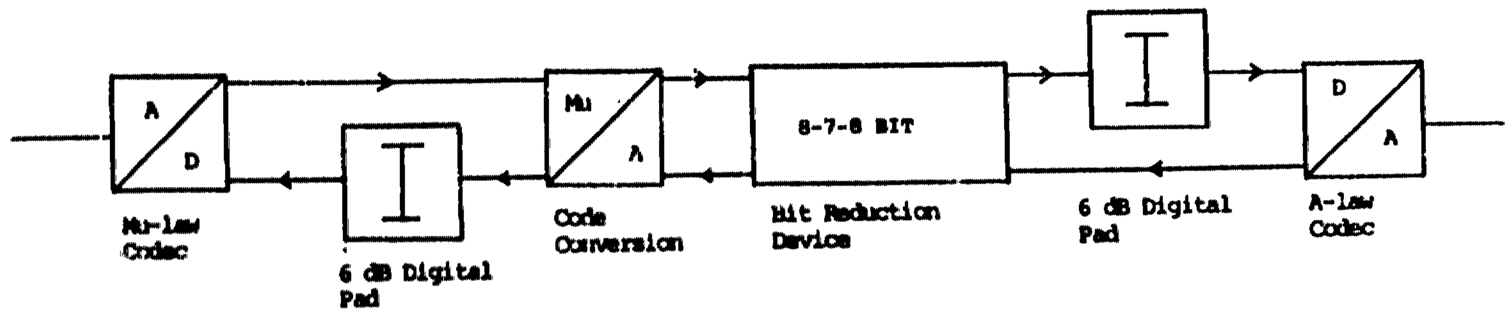


Figure 2b - International connection with 6dB digital pad and bit reduction device

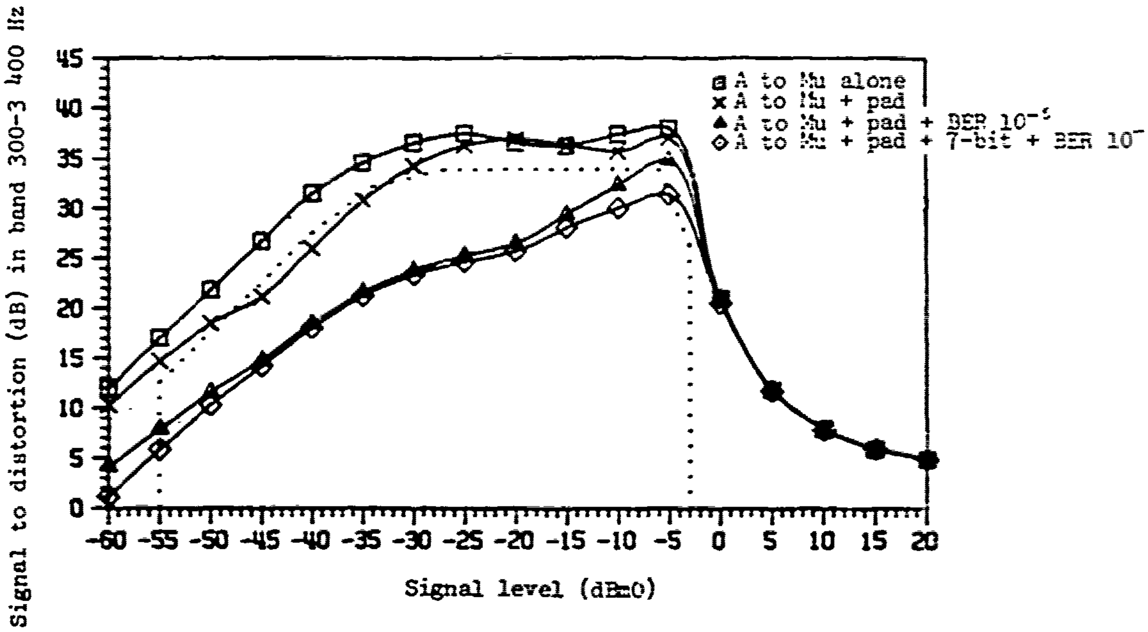


Figure 3a - Effect of tandem devices (A to Mu)

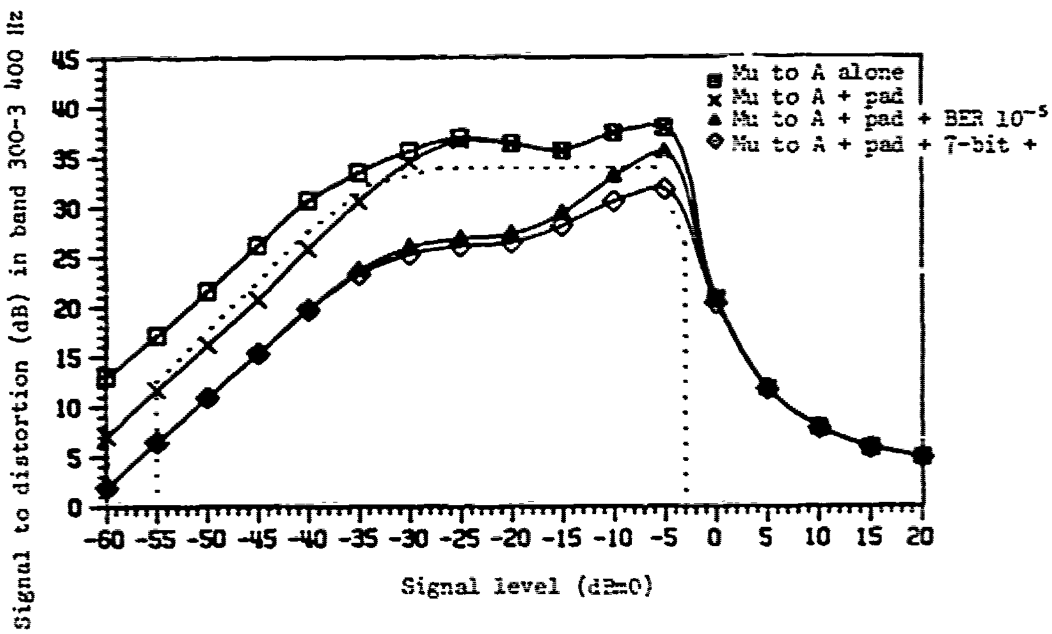


Figure 3b - Effect of tandem devices (Mu to A)

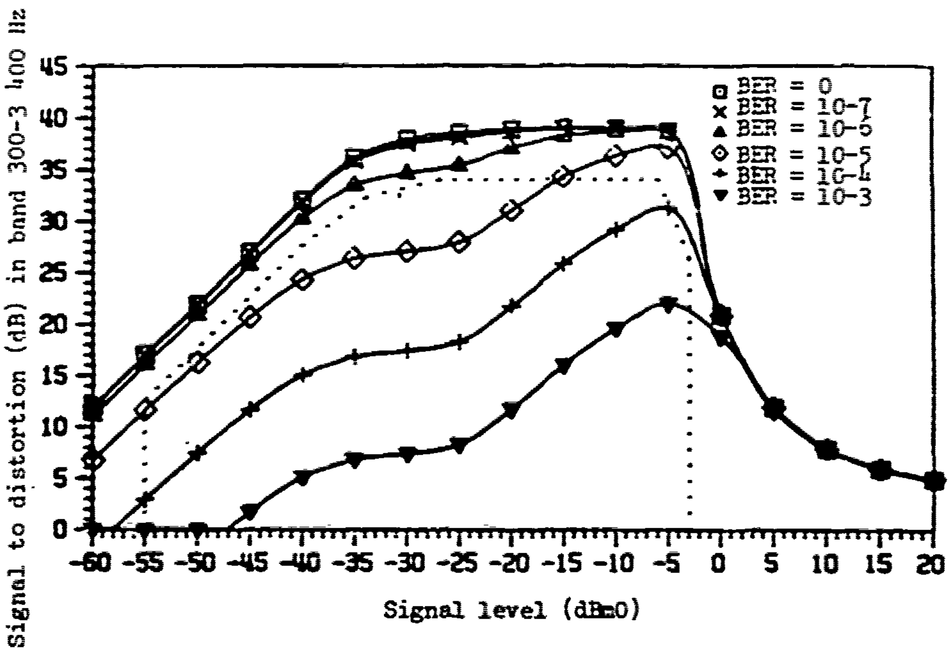


Figure 4a - Effect of random bit errors on ideal 8-bit A-law PCM

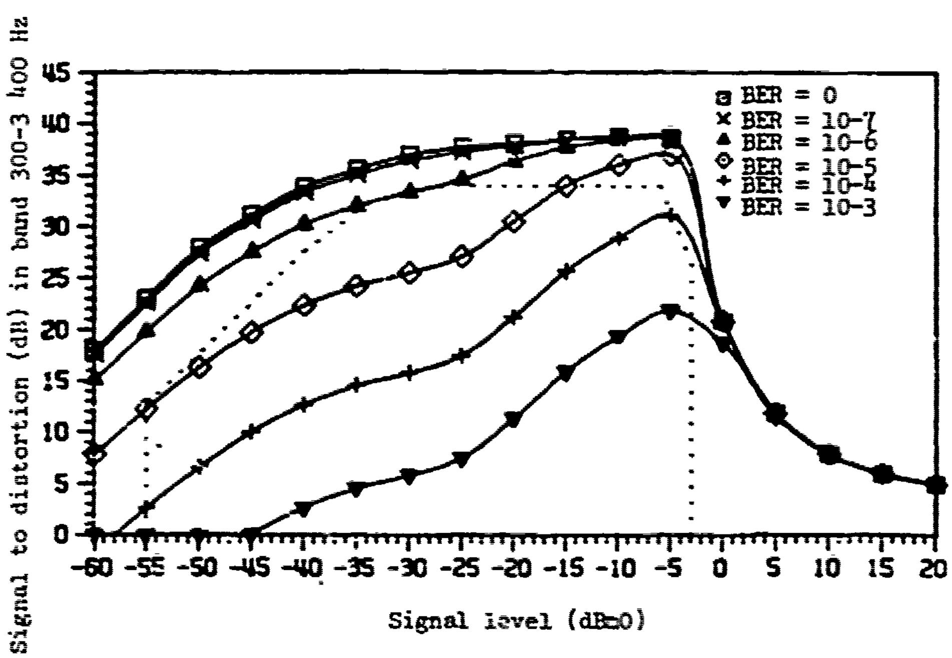


Figure 4b - Effect of random bit errors on ideal 8-bit Mu-law PCM

QUESTION 10/XVIII - Availability for the ISDN

(continuation of Question 5/XVIII, studied in 1977-1980)

1. What availability parameters should be defined for the ISDN, and what values should be recommended for each parameter ?

In particular, values of parameters should be defined on an overall basis (customer to customer) and apportioned as appropriate to nodes and links in the network for different aspects of each service (see Annex).

2. What methods should be used to measure the availability ?

3. Which hypothetical reference models should be coordinated with the studies on determination ?

Note 1 : Studies of this Question should be based on the results of those carried under Question 9/XVIII and in close cooperation with the appropriate Study Groups.

Note 2 : Replies to this Question will be transmitted to Study Group CMBD as contribution to Question 2/CMBD by the Rapporteur for availability.

Note 3 : Meeting these aims requires careful study of, an [illegible]ed approach to, all aspects of service availability such that particular parameters are not considered in isolation.

Annex

(to Question 10/XVIII)

Reply to Question 5/XVIII (Reliability and availability of digital networks), Study Period 1977-1980

General aspects

The availability and reliability performance objectives are highly dependent on the definitions of failures in a network. These are different for different services and one failure also has different effects on the functions of different services. Since the various availability requirements depending on different services, leads to different costs in the network, it is necessary to start with at least one objective for each service and then try to combine them in such a way that the number of objectives will be reduced.

As a basis for further studies at least the following four levels of performance will be used :

1) **Normal service**
2) **Degraded for data**
3) **Interruption for non telephony**
4) **Total interruption.**

These levels as well as the objectives have to be defined with a number of suitable parameters and the study of which parameters can be used is a task for the next study period.

The service availability objective will be established taking into consideration the quality of service offered to the subscriber and that which can be achieved by the Administrations. The Administrations will be able to allocate their maintenance personnel and procedures and the provisioning of stand by equipment and alternative circuits in a manner most appropriate to themselves but at the same time achieving the performance objectives.

Calculating failure occurrences and the probability for a failure in the network makes it necessary to use statistical methods. This should be taken into account when evaluating or measuring failures in a network.

From the above is understood that specifying all failures and failure effects in a complete network with a number of different services is a rather complicated task. This makes it necessary to use simplified models, such as hypothetical reference circuits and the work with those must continue with high priority during the next studyperiod. One first model has been developed during the study period (see Figure 1).

When an overall availability performance objective has been formulated, the next very important problem is how to allocate values (requirements) to the various parts of a network. This is an economic problem of high importance and possible optimization methods must be dealt with as soon as possible.

Conclusions and future work

a) The principle to divide the network into two basic parts, (1) subscriber sub-system, (2) linking sub-system has been provisionally adopted as the basis for further studies. It was also agreed that the linking sub-system portion should receive initially the greatest emphasis. This principle is described in more detail in Appendix.

b) A simplified calculation model should be used. Figure 1 is such a model and represents a part of a switched connection in an integrated digital network. This model does not include alternate routing or rerouting. Sufficient redundancy may be included to achieve the desired level of availability, considering also maintainability.

c) This model could later be expanded to include 2 or more routes between the two switching centres and the achievable availability under these conditions may be determined. It may be desirable to indicate such objectives in both of the following manners : (1) at any instant of time x per cent of the paths shall be available, and (2) any particular path should be available y per cent of the time.

Additional refinements to the model may also be achieved by including alternate routing through other switching centres and network management principles. These refinements may indicate that a desired availability objective may be achieved for the linking sub-system with lower values of availability objectives for individual parts of that system than originally indicated by the study of the basic model proposed in (b). In all cases, it has been assumed that sufficient paths have been provided using traffic engineering principles to obtain a specified grade of service.

The interdependency of traffic engineering and path availability objectives must finally be determined.

d) Availability objectives for the subscriber sub-system would be studied in the future or left to the responsibility of each national network Administration.

e) The definitions of faults and the effect of a fault is fundamen'al for the availability and reliability of a network. Definitions such as error characterizations and data (values) are needed. During this study period the following values have been stated :

- An error ratio in excess of 1.10^{-3} is generally regarded as being a criterion for unavailability.

- On a 64 kbit/s path this may more conveniently be expressed as an error count of more than 64 errors per second, persisting for x seconds.

- As a time criterion for unavailability 1...2 seconds have been mentioned initially for digital service, as this is the time after which digital multiplexes equipments become normally unavailable when detecting an exc sible error rate. A value of 10 seconds for telephony, as proposed by Study G X. has been considered. The exact value should be decided in cooperatic h relevant Study Groups (such as Study Group XI and CCIR Study Groups).

f) The description of the availability concept and other definitions being studied by Joint Study Group CMBD will be used as the basis of availability work in Study Group XVIII.

g) Different levels of performance have to be taken into account in an ISDN. Together with the probability for a subscriber to notice the different levels this is essential for the future studies. More information and some proposed values are given in Annex 5 of Question 9/XVIII.

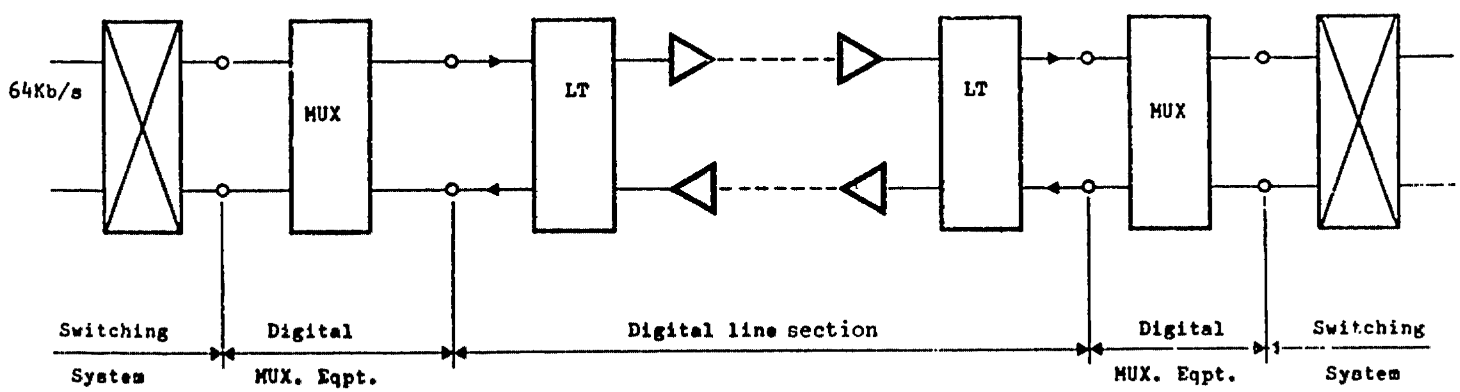


Figure 1 - The simplified calculation model of a part of a digital connection for availability studies

LT - Line Terminating Unit

Appendix

(to Annex to Question 10/XVIII)

Availability of digital transmission systems

(contribution by the Italian Administration)

1. Introduction

With reference to Question 5/XVIII in this contribution, the Italian Administration presents some general considerations in order to define, evaluate and express the availability of digital transmission systems.

2. Networks and transmission systems

An analogue or digital telecommunication network can be subdivided into two basic sub-systems :

1) subscriber sub-system - that includes all the parts being assigned to subscribers and allowing the access to the network (telephone set, individual line connecting the subscriber with the local exchange, local exchanges);

2) linking sub-system - that includes the plants and the facilities in common among all the users which are assigned on demand by a frequency division (analogue transmission), space division (space division switching), or time division (digital transmission and switching).

Aiming to a study concerning the interconnection and the interworking of systems belonging to different Administrations, it is mainly important the linking sub-system which here means Integrated Digital Network (IDN) and Integrated Service Digital Network (ISDN).

Such a network includes :

a) nodes - where switching, signalling, multiplexing and A/D conversion are carried out;

b) branches - links connecting the nodes in different ways.

As a first approach to the problem, the study of the transmission systems on the reliability point of view could be carried out : they actually correspond to the branches and partially to the nodes (the exchanges are excluded).

In fact, the transmission systems are defined as the whole of the transmission facilities that fulfil bi-directional paths suitable for transmitting useful telecommunication signals between two terminals.

The transmission system is generally subdivided into two systems :

a) multiplex sub-system - that fulfils the A/D conversion of the signal and the multiplexing at different hierarchical levels;

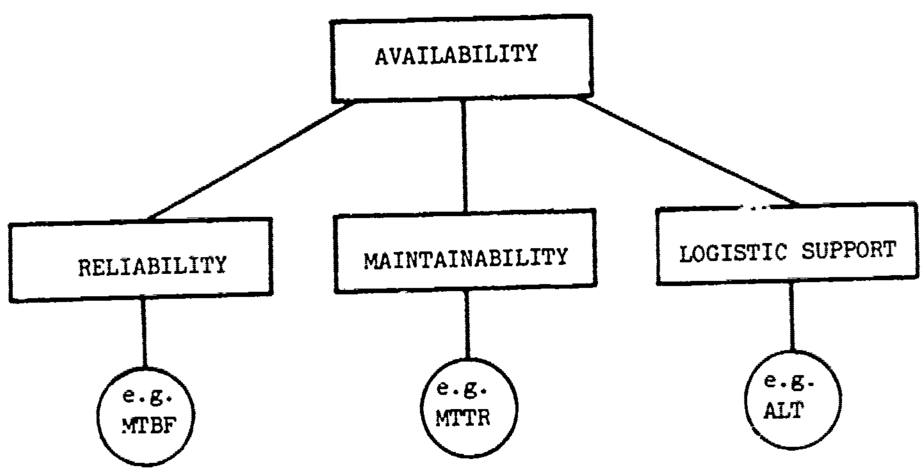
b) line sub-system - that fulfils the bi-directional transmission over physical lines (cables, guides, fibres), including them as well as the regenerators.

Because of the complexity of the telecommunication networks and the variety of the transmission systems, some peculiar characteristics of them, useful for determining a reference model, should be chosen.

3. Availability of networks and systems

Considering the telecommunication network, it is important to take into account the operating and service point of view, therefore, the availability concept seems more interesting than the reliability aspect when applying to the networks.

The availability, in a large sense, includes the aspects of the reliability and the maintenance concerning the maintainability of the systems as well as the logistic support of maintenance (Figure 1).



MTBF - MEAN TIME BETWEEN FAILURES (REPAIRED SYSTEMS)

MTTR - MEAN TIME TO REPAIR

ALT - AVERAGE LOGISTIC TIME

Figure 1

So, it appears as a combined concept where the system and the service aspects are included. But no availability objective has been recommended by CCITT as concerns the network : only recently CCIR has fixed an objective concerning the radio-relay systems.*)

This objective can be far from the value relating to the networks, because in some cases very high availability values, valid for the networks, could be met using re-routing techniques of large assemblies of circuits instead of systems having high intrinsic availability.

Really, a complete study concerning the availability of the networks should also include the traffic aspects and should take into account the dynamic management criteria (e.g. re-routing techniques).

4. Availability of paths

The complexity and the modern management of the networks, based on the dynamic operating, also suggests to consider the availability of paths instead of the availability of systems.

So, in addition to the maintenance aspects (in a large sense) covered by the availability concept, it is possible to take into account the structure of the network, i.e. the network redundancies, the distribution of circuits between different transmission systems, the influence of sub-systems outside the transmission system under consideration but indispensable to its operation (e.g. no-break power facilities), and the manual or automatic re-routing.

In fact, it should be noted that an interruption, due to failures or maintenance operations concerning an individual circuit or a circuit assembly, may not mean a consequent interruption of traffic if suitable re-routings have been carried out according to agreed procedures corresponding to network planning criteria.

As a first approach to the study of Question 5/XVIII, the traffic and dynamic management aspects have not been taken into account and a simplified model could be developed on the basis of this assumption.

5. Conclusions

In this contribution, attention is drawn on the complexity of the study relating to Question 5/XVIII. As a first approach, the strong assumption of negliging the traffic concept is proposed in order to develop a reference model suitable to define and evaluate the availability of paths, taking into account reliability as well as the maintenance aspects.

*) Recommendation No. 557 (Volume IX, Kyoto, 1978).

QUESTION 11/XVIII - Characteristics for digital sections

(Continuation of part of Question 11/XVIII, studied in 1977-1980)

a) What essential common criteria should be established for all types of digital line sections and digital radio sections ?

Note 1 : Consideration should be given to the need for digital line sections and digital radio sections to be interchangeable and interconnectable. However significant differences may exist between performance characteristics for systems using different media.

Note 2 : Coordination with CCIR Study Groups 4 and 9 must be undertaken to establish the common criteria.

b) In which way should the existing Recommendations of the G.9xy series be amended and completed, insofar as they relate to digital sections ?

c) What new Recommendations should be established regarding digital sections (e.g. using non-hierarchical bit rates) ?

d) What are the principles which should form the basis for the detailed study of the local network digital line transmission systems and multiplexers for connecting digital terminals to the ISDN ? (To include combination of analogue and digital terminals and PABXs).

For each individual type of digital line section, the following specific points require study :

- bit rate;
- special properties (such as bit sequence independence, or restrictions of the bit sequence that may be transmitted);
- characteristics of interfaces (normally these should be in accordance with Recommendation G.703);
- error performance (expected to comply with Recommendation G.821 which specifies the overall network performance);
- jitter performance (input and output jitter as well as jitter transfer function; Recommendation G.703 should be observed);
- other performance parameters;
- availability;
- fault conditions and consequent actions.

Annex

(to Question 11/XVIII)

Criteria for the fault condition "error ratio 1.10^{-3} in digital line sections at 2048 kbit/s

(Contribution from Federal Republic of Germany)

1 Introduction

The fault condition "Error ratio 1.10^{-3}" in digital line sections at 2048 kbit/s corresponds to the fault condition "Excessive error rate" in 2048 kbit/s primary multiplex equipment, which is specified in detail in Rec. G. 732. With respect to the important consequent actions (prompt maintenance alarm, emission of AIS), it is necessary to define a similar detailed specification for the fault condition "Error ratio 1.10^{-3}" in line sections at 2048 kbit/s.

The fault condition "Error ratio 1.10^{-5}" is less important since it only implies a deferred maintenance alarm. Its detailed specification can therefore be left to the national Administrations.

2 Modifications with respect to Rec. G.732

Error detecting method

In primary multiplex equipment, errors are detected in the frame alignment signal; in digital line sections, errors are detected by code rule violations. The relation between violation rate and bit error rate depends on the line code, the choice of code rule violations to be detected by the error detector, and on the binary signal pattern.

In actual operation, the statistical properties of the binary signal pattern may be described by two limiting cases. If all channels are busy, the binary signal approaches a random pattern; if all channels are idle, the binary signal approaches a 1010...pattern; if only part of the channels is idle, the statistical properties of the binary signal are somewhere between these limiting cases.

In order to get a clear definition of the probabilities of activating and deactivating the indication of fault condition, it is necessary to define a reference pattern for which the specification should be met. It is suggested to select a $2^{15}-1$ pseudo-random pattern according to Recommendation O.151.

Measuring time for activating the indication of fault condition

The code violation frequency is in any case by at least one order of magnitude higher than the frequency of errors in the frame alignment signal. Therefore, the measuring time for activating the indication of fault condition can be reduced to a few tenths of a second as compared to the "few seconds" in G.732.

Insensitivity to error bursts

The error rate detector in a line section should have the same insensitivity to error bursts as the error rate detector and the loss of frame alignment detector of primary multiplex equipment. In primary multiplex equipment, an error burst of up to four frame lengths ($\leq$ 0,5 ms) does not activate any indication of fault condition since only two errors in the frame alignment signal are detected.

Criteria for deactivating the indication of fault condition

The definition of the criteria for deactivating the indication of fault condition must take into account the argument concerning the mutual dependence between error rate, service alarm and binary signal pattern, which is indicated in Doc. XVIII-No.217, Appendix 4 to Annex 1. When the fault condition "Error ratio 1.10^{-3}" is detected in a digital line section, the digital path is taken out of service and the binary digital signal pattern is changed into the idle pattern (..10101...) of the multiplex equipment. As a consequence of this change of pattern, the error rate may decrease considerably and the indication of fault condition "Error ratio 1.10^{-3} in the digital line section may be deactivated, unless the threshold for deactivation is sufficiently low. In accordance with the above-mentioned document, it is proposed to deactivate the indication of fault condition not before the error rate has fallen below 1.10^{-5}.

3. Proposed formulation of G.912, Section I.4.1.3

I.4.1.3 Error ratio 1.10^{-3} detected by code rule violations.

I.4.1.3.1 Criteria for activating the indication of fault condition:

- **Error** ratio $\leq 1.10^{-4}$
  The probability of activating the indication of fault condition in a few tenths of a second should be less than 10^{-6}.
- **Error** ratio $\geq 1.10^{-3}$
  The probability of activating the indication of fault condition in a few tenths of a second should be higher than 0.95.

The indication of fault condition should not be activated by an error burst $\leq$ 0,5 ms.

I.4.1.3.2. Criteria for deactivating the indication of fault condition:

- **Error** ratio $\geq 1.10^{-3}$
  The probability of deactivating the indication of fault condition in a few seconds should be almost 0.
- **Error** ratio $\geq 1.10^{-5}$
  The probability of deactivating the indication of fault condition in a few seconds should be less than 0.05.
- **Error** ratio $\leq 1.10^{-6}$
  The probability of deactivating the indication of fault condition in a few seconds should be higher than 0.95.

Note: The criteria are valid for a $2^{15}-1$ pseudo-random pattern according to Rec. O.151.

QUESTION 12/XVIII - Maintenance philosophy of the digital network (continuation of part of Question [illegible]/XVIII, studied in 1977-1980, of interest to Study Group IV)

Considering

a) that work carried out in the definition stage of the maintenance philosophy during the Study Period (1977-1980) has included a number of the implementation aspects of the philosophy (Recommendation G704);

b) that a well defined maintenance philosophy will determine the direction of future maintenance studies;

c) that Recommendation G704 is incomplete (e.g. no digital switching and signalling considerations);

d) that certain operational aspects of network and traffic management may influence maintenance philosophy;

e) that implementation of the philosophy is complex and requires separate treatment;

What is the overall maintenance philosophy for digital networks ?

The following specific points require study :

1. What additional principles are needed to ensure that the maintenance philosophy encompasses all elements of the network ?

2. To what degree is network surveillance required to identify status and quality of connections and network elements ?

3. What is the effect of differing service requirements on network maintenance philosophy ?

Annex

(to Question 12/XVIII)

Test sequence to measure the bit error rate on 64 kbit/s channels

1. Introduction

The transmission of digital information at a rate of 64 kbit/s is possible via digital PCM transmission systems as well as via (analogue) carrier frequency systems.

The interface for the digital 64 kbit/s signal is specified in Recommendations G.703 and G.732. In Figure 4/G.702 the 64 kbit/s digital path is illustrated. The hypothetical reference circuit is described in Recommendation G.721.

The modem according to Recommendation V.36 allows the transmission of a 64 kbit/s signal in a primary group in the frequency band 60 to 108 kHz via the analogue carrier frequency transmission system.

For bit error rate measurements on the above channels and their combinations the specification of a standard test sequence is urgently required.

2. **Proposal**

It is proposed to specify a pseudo-random pattern as a test sequence which has a pattern length of $2^{11}-1 = 2047$ bits. This test sequence can be produced by means of an 11-stage shift register with feedback from the outputs of the 9th and 11th stage of the first stage via an exclusive OR gate.

It should be noted that this test sequence contains a maximum of 10 consecutive "0" bits. Therefore in the case of international testing where the measurement includes systems based on 1544 kbit/s it is necessary to modify the test sequence in such a way to avoid more than 7 consecutive "0" bits. The specific details of implementing this approach is left to Study Group IV.

Two essential conditions are met by the proposed test sequence :

a) The test sequence is a maximum run length pseudo random sequence, (which means that its generating polynomial is prime and primitive), and if the number of stages of the shift registers in the scrambler is less than that of the test sequence generator, then the scrambler and the test-sequence generator cannot have a common factor. In such case there will be no restrictions with respect to possible scrambler configuration.

b) The period of the test pattern does not exceed the time still convenient for practical measurements. On the other hand, the test sequence is sufficiently long to closely simulate the random signal being present in practice.

In addition to the random pattern capability described above, this test equipment should provide for fixed patterns. The definition of these patterns is under study.

Since the proposed test sequence can also be used at bit rates of 48 kbit/s to 72 kbit/s, it should be considered whether in view of the advantages mentioned above, it would be advisable to replace the test sequence described in Recommendation V.57 by the pattern of $2^{11}-1$ bit length. In any case, this pattern should be mentioned in V.57 as a possible alternative.

It should be noted that Study Group XVII has followed the proposed test pattern of $2^{11}-1$ length.

QUESTION 13/XVIII - Implementation of maintenance philosophy
(continution of part of Question 4/XVIII, studied in 1977-1980)

1. Considering

a) that parameters indicating network performance have only been defined in a preliminary manner, more work will be required to examine their limits for network maintenance purposes;

b) that maintenance techniques and procedures will be required for all elements of the digital network;

c) that network and system testing procedures should be recommended to enable the provisioning of testing and diagnostic capabilities in systems and equipment;

d) that there is a need to provide coordinated, unambiguous alarms and indications to efficiently isolate failed network components and restore service;

e) that the concept of maintenance entities and sub-entities needs to be further defined;

f) that it is necessary to ensure a common interpretation of the maintenance philosophy by all Study Groups that are concerned with the evolution of the network;

g) that there is a need to inform other Study Groups of the maintenance implementation strategies that have been applied to specific equipment Recommendations;

1.1 How should the Maintenance Philosophy be implemented in digital networks and coordinated with the work of other Study Groups?

The following specific points require study:

i) What are the maintenance parameters, their limits, measurement methods and their relationship to network performance?

ii) How to ensure performance compatibility for thresholds and operate times among service alarms, maintenance alarms and protection switching?

iii) What considerations should govern the location and design of measuring equipment to facilitate overall network maintenance and operation?

iv) What specific network surveillance capability should be made available for network operation and management?

v) What further considerations should be given to the implementation of the principles of maintenance entities and/or sub-entities?

Note 1: Recommendations for measuring equipments not included in the digital equipment will be made by Study Group IV, taking into account the results achieved by Study Group XVIII under this Question.

Note 2: At some future date much of the detailed work under this Question should be undertaken by other Study Groups (e.g. IV, VII, XI) although this may not be appropriate at the present time.

QUESTION 14/XVIII - Interworking between digital systems based on different standards

(Continuation of Question 14/XVIII, studied in 1977-1980)

What measures are required and what recommendations have to be made to enable interworking between digital systems based on different standards ?

The following specific points require study :

Point a) - Conversion between different encoding laws in primary PCM multiplex equipment (as specified in Recommendation G.711) taking into account the possible use of 64 kbit/s paths for signals other than telephony.

Point b) - Conversion between different frame structures of primary PCM multiplex equipment (as specified in Recommendations G.732 and G.733) and between higher order multiplex equipment (as specified in Recommendations G.742, G.743 and so forth).

Note 1 : When studying this Question, priority of consideration should be given to interconnections over international satellite links.

Note 2 : In undertaking this study, there are many detailed questions which must be addressed (see Annex 1). The results of the studies undertaken in the 1977-1980 study period should be considered (see Annexes 2 and 3). Close cooperation is required with CCIR Study Group 4 (see Annex 1) and with CCITT Study Groups VII, XI, XV and XVII (see Annexes 2 and 4).

Annex

(to Question 14/XVIII)

Detailed Questions for further study

The following list of questions includes those raised in the 1977-1980 Study Period, since all items were continued for further study. New items have also been added, as identified in that study period.

These questions are based on the satellite communication link layout on Figure 1, as well as on the definitions of Type I and Type II Satellite Systems. These system types are:

I. A system wherein a 1544 kbit/s or a 2048 kbit/s signal is in essence carried transparently to the other end of the satellite link without processing of bits internal to these bit streams. This type of system may not include Time Division Multiple Access (TDMA) function, or it may include a TDMA function transmitting CCITT standard (e.g., 1544 or 2048 kbit/s) signals.

II. A system wherein 1544 kbit/s and/or 2048 kbit/s signals are subjected to processing demultiplexing the primary multiplex signals. The multiplex conversion function is expected to be performed at one or both Direct Digital Interface Equipments (DDIEs).

Some of the questions require input from other CCITT study groups and from CCIR. Initial queries to these groups were made in 1979 (CCITT) and 1980 (CCIR).

A. Questions relating to Type I Systems

i) Where should the MSC be located, and why?

a) A-law country. μ-law country, or either?

b) International exchange or earth station site?

ii) What is an appropriate MSC capacity?

a) Is a 24-channel to 30-channel MSC (i.e., a single system of each standard) needed? If so, should the remaining six channels in the 30-channel system be unused?

b) What combinations of primary multiplex levels should be provided (e.g., 120 channels representing four 30-channel systems and five 24-channel systems)?

c) Are any nonhierarchical MSCs needed ?

iii) Where should A-law to from μ-law conversion be handled in the μ-law country ?

a) In the MSC or externally ?

b) At the international exchange or earth station site?

iv) What synchronization requirements are to be considered when using the MSC?

v) Should it be possible to include a digital speech interpolation (DSI) function in Type I systems? If so, where should this function be added? What answers (if any) change as a result of adding this function?

B. Questions relating to Type II Systems. (Answers may vary depending upon particular Time Division Multiple Access (TDMA) plans.)

i) Are there any special multiplex conversion problems when using Type II Systems? If so, what are these? Are Recommendations required?

ii) Should A-law to/from μ-law conversion be done in the earth station equipment or in the international exchange in the μ-law country?

iii) Do any special synchronization problems arise in the CCITT recommended systems? If so, what are these problems, and how might they be solved?

iv) Should it be possible to include a digital speech interpolation (DSI) function in Type II systems? If so, where should this function be added? What answers (if any) change as a result of adding this function?

C. <u>General questions</u>

i) What signalling means will be utilized to determine that particular channels are carrying voice, voiceband data, or digitally generated data service? (Note: this information is needed wherever the A-law to/from μ-law conversion is to take place, as well as whenever other special actions must be taken - see the next three questions.) Should these signalling systems be either common channel or channel-associated, or a combination? If channel associated, should analogue (e.g., PCM-encoded sinusoidal signals) means be included?

ii) What other processing may be needed for channels carrying voice services in these systems (e.g., echo suppression or cancellation)?

iii) What special measures may be needed for channels carrying voiceband data services?

iv) What transformations may be needed for channels carrying digitally generated data services?

v) What transformations may be needed for channels carrying CCITT-recommended signalling systems signals?

vi) Will other methods of encoding than standard A-law or μ-law PCM be utilized for voiceband services on satellite links? If so, what impact will these have on the system and on the answers to other questions?

vii) What alarms and related system information should be transferred across the interface? What algorithm should be used? What action should be taken at the earth station and/or at the international exchange with alarms generated at the opposite end of the link? (Note: 2048 kbit/s systems can transmit alarms in time-slots 0 and 16.)

viii) What special maintenance approaches are appropriate for the systems postulated? Should the satellite and the two terrestrial links be considered separately for maintenance purposes?

ix) Should n×64 kbit/s services be carried? If so, what special problems arise and how may they be solved?

x) Are there known interworking incompatibilities between any user services (e.g., direct digital data) specified or contemplated for the two hierarchies? If so, what are these and how might the incompatibilities be resolved?

xi) Should 31-channel versions of the 2048 kbit/s system (i.e., systems using time slot 16 for service) be considered? If so, what changes may be needed in the answers to the previous questions?

xii) In some situations, a country using one primary multiplex standard may receive via satellite signals encoded in a format nonstandard in that country, transport these signals (perhaps at considerable distance, and perhaps through international exchanges) to a second earth station, and retransmit these signals to a third country using the original format. In these situations, should recoding be allowed? If not, what arrangements should be considered for providing this service? What, if any, is the impact on the answers to other questions?

xiii) With some choices of answers to the previous questions, a terrestrial link (Figure 1) may either require a 2048 kbit/s system in a 1544 kbit/s country (or vice versa), or require a nonstandard system (such as A-law encoding - with zero suppression difficulties - on 1544 kbit/s facilities). Should such facilities be allowed? If so, how should they be specified? Should they be standardized for terrestrial link use?

Note : Answering some of these Questions requires interaction with CCIR. To further this process, a communication was sent from CCITT Study Group XVIII to CCIR Study Group 4 at the end of the 1977-1980 study period. The text of this communication is appended to this Annex.

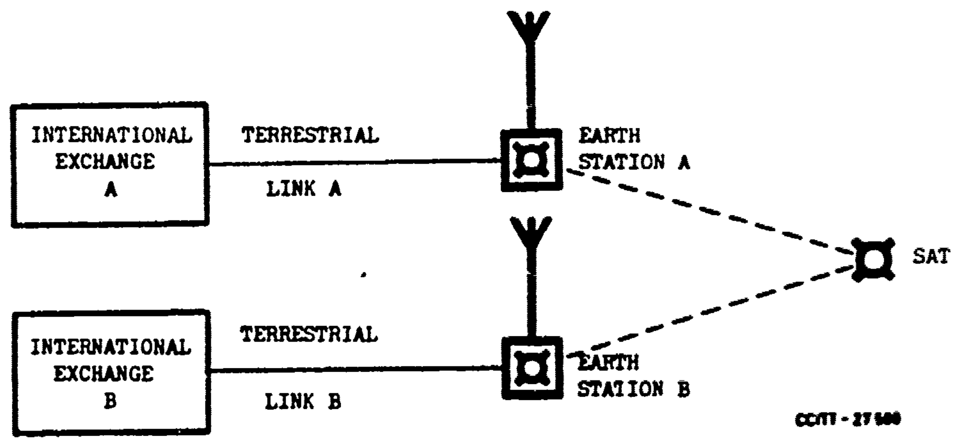


Figure 1 - General layout of satellite communication link

Appendix

(to Annex 1)

Communication from CCITT Study Group XVIII to CCIR Study Group 4

Whereas CCIR, at its XIVth Plenary Assembly in Kyoto 1978, adopted Opinion 57 and Report 707, with portions of these documents related to international satellite links involving the interworking of countries using the 1544 and 2048 kbit/s hierarchies, CCITT Study Group XVIII asks CCIR Study Group 4 :

1. To inform CCITT Study Group XVIII whether DDIE equipment (specified by CCIR; see e.g. Opinion 57) will always include the function of reassembling the primary multiplex signal used in the receiving country (as implied by Sections 3.1 and 3.2 of Report 707), or whether there are also contemplated systems wherein a 1544 kbit/s or a 2048 kbit/s signal is in essence carried transparently through the satellite link without processing of bits internal to the primary multiplex bit stream, with multiplex system conversion to be performed by equipment specified by CCITT.

2. To inform CCITT Study Group XVIII whether digital speech interpolation (DSI) functions are likely to be incorporated and, if so, whether the DSI function will be incorporated in the direct digital interface equipment (DDIE), in equipment which will be located on the side of interface "A" (see CCIR Opinion 56) containing equipment specified by CCITT, or on either side of interface "A" depending upon the application.

3. To consider the implications of the answers to points 1 and 2 with regard to the ultimate need to specify similar functions on both sides of interface "A" for different applications and, if such a need is perceived, to suggest mechanisms whereby the recommendations developed by CCIR and by CCITT may be kept consistent.

4. To inform CCITT Study Group XVIII if any systems are postulated in which CCIR specified equipment will modify the code of PCM-encoded voice signals and, if so, of the nature of the new code and of any impairments expected to voice and voiceband data signals.

5. To inform CCITT Study Group XVIII if there is any contemplated difficulty in meeting the slip performance for plesiochronous interworking as recommended in Recommendation G.822. It is expected that, in order to meet the requirements of Recommendation G.822, high-accuracy clocks in accordance with Recommendation G.811 will be required.

Annex 2

(to Question 14/XVIII)

Summary of interworking studies during the 1977-1980 study period

Study during 1977-1980 of interworking between two countries using different primary multiplex standards concentrated on satellite applications. Most contributions responded to some portion of the detailed questions raised by the Rapporteur. In the discussion below, reference is made to the corresponding questions carried over; see Annex.

Responses to the basic questions regarding location of the A/μ law converters and of the multiplex system converters are summarized in Table 1. With regard to the remaining questions, the following comments may be made:

Question A ii): Several contributions mentioned Multiplex System Converter (MSC) capacity. The proposals ranged from inefficient 24-30 channel system interfaces (at least for early satellite systems or those with small cross-sections) to 120-channel (5 × 24 and 4 × 30) interfaces, to the possibility of future higher cross-section interfaces.

Question A iv): The contributions suggested no special synchronization requirements when the countries connected utilize synchronous national networks; interworking is then plesiochronous. In the case wherein slip type DDIEs are employed in a satellite system which is timed to clocks of low accuracy, high slip rates will probably result. A CCIR report described general requirements on buffers associated with system interfaces. Both CCIR and KDD mentioned the functions of justification and slip-type DDIEs.

Discussion on this point emphasized the preference of CCITT Study Group XVIII that interworking between digital terrestrial and satellite links be plesiochronous, using high accuracy clocks to provide satellite TDMA timing. Some Administrations noted that satellite earth station equipments may have difficulty gaining access to highly accurate national clocks and may not be able to afford their own highly accurate clocks.

Question B i): No particular problems with implementing Type II systems were identified.

Question B iii) : Synchronization comments are similar to those in Question A iv).

Question C i) : The only substantive input regarding signalling was received from Study Group XI, which indicated that, in accordance with Recommendations Q.7 and Q.110, CCITT Signalling Systems Nos. 5, 6 and 7 and R1 and R2 can be operated with circuits including satellite links. Of these, only signalling systems Nos. 6 and 7 can offer the required additional signalling capacity for meeting the requirements imposed by alternate voice and data applications; however, these additional functions do not yet appear in the existing Recommendations, Study Group XI has proposed two Questions (Q.2/XI and Q.3/XI) for addressing these issues for signalling system No. 7 (which according to Recommendation Q.7 is the preferred system for interexchange signalling in the IDN and ISDN) in the 1981-1984 study period.

Study Group XVII also noted the importance of providing a means of differentiating between the various types of ISDN services to control telephone ancilliary equipment when interworking between ISDNs based on different PCM/TDM standards.

Question C ii): **Other types of processing that may be needed for channels carrying voice systems have been identified, including:**

- **echo suppression or cancellation**
- **digital speech interpolation**
- **inversion of bits 3, 5, 7 (J transformation of Figure 1, Annex 3); substitution of 00000001 for 00000000 (Z operation of Figure 1, Annex 3)**

Questions C iii) and C iv): **Other types of processing that may be needed for channels carrying nonvoice systems have been identified, including:**

- **substitution of 00000001 for 00000000 (Z operation)**
- **inversion of bits 3, 5, 7 (J transformation)**
- **inversion of all bits (I transformation of Figure 1, Annex 3) and/or inversion of bits 3, 5, 7 (J transformation) for CCITT Signalling System No. 7**
- **possible digital processing (SG VII comment)**

Voiceband data and direct digital data both may need processing different from voice and different from each oth. Refer to Annex 4, which identifies Study Group VII concerns and specific questions in this area.

In addition, it was noted that digital speech interpolation equipment, if used, may alter the answers to some of the questions.

It is clear that close cooperation is needed between Study Groups VII, XI, XVII and XVIII to allow progress in the study of the interworking question.

TABLE 1

Summary of inputs

Input / Topic	COMSAT	AT&T	Teleglobe	KDD	CCIR	Study Group XI
Location of A/μ converter						
A. Country						
- μ-law (per G.711)	X	X	X	X		X
B. Site						
- earth station	X			Note 3		
- transit centre				X		X
- either		X	X			
Location of MSC						
A. Country						
- A-law						
- μ-law	X		Note 1			X
- either or both		X	Note 2	X	X	
B. Site						
- earth station	X	X		X		
- transit centre				Note 4		X
- either			X			
C. Functional integration						
- into equipment covered by CCITT Recommendations						X
- into equipment covered by CCIR Recommendations						
- into equipment covered by either CCITT or CCIR Recommendations	X			X	X	

Note 1 : The MSC should be located in the μ-law country if the satellite system transmits a fully standard A-law, 2048 kbit/s signal as specified in Recommendations G.711 and G.732.

Note 2 : The MSC function could be located in either or both countries if the satellite system transmits a signal not meeting the constraints of Note 1.

Note 3 : The A/μ converter could be located in the earth station if DSI equipment is also located at the earth station.

Note 4 : [illegible] specific transit centre.

Annex 3

(to Question 14/XVIII)

Interworking between two standards considering voice and data

1. Introduction

This Annex is an example of an approach meeting some of the interworking problems. As such, it is considered worthy of further study, along with other possit solutions which may be proposed. In particular, it is important to note that Study Group XI has requested that Study Group XVIII search for a solution to the interworking problem which accomplishes all necessary transformations at one location.

In the remainder of this annex, one possible interworking configuration between an A-law country and a μ-law country is investigated. A new method with a single code conversion is introduced to solve the basic problem of applying the A/μ conversion to voice signals but not to digital data signals, while suppressing the consecutive all-zero pattern on 1544 kbit/s transmission links.

2. Requirements

The following items need to be considered :

1 For digital data signals, the A/μ and μ/A conversions must be removed, because the conversions are not uniquely reversible.

2. The all-zero word (8 bits) may be inhibited on 1544 kbit/s transmission links.

3. The A/μ converter may be located in the μ-law country and its preferable location is the gateway switch (except in the case that the DSI, which needs voice channel identification, would be located at the earth station).

4. The MSC (Multiplex System Converter), which may be located at the earth station of either the A-law or μ-law country, should preferably perform the same conversion without distinguishing between voice and data.

5. The deficiency caused by the interworking shall be minimized, although it can not be avoided perfectly because of the all-zero restriction.

3. A/μ converter

Before detailed discussions the bit inversion process related to the A/μ conversion should be clarified.

It is understood that the so-called even bit inversion operation is for pur descriptive purposes in the CCITT Recommendations. It is only a logical process wit the A-law codec. (See Note 2 of Tables 1a and 1b in Recommendation G.711, and Note : of Section 1.1 in Recommendation G.732.) Therefore, it is taken for granted in th following considerations that this even bit inversion is included in the A/μ converter as an internal logic.

4. Interworking

Feature : Introduction of the Z operation (by which the code "00000000" is replaced by "00000001" but others are not changed) to avoid the transmission of all-zero word on 1544 kbit/s transmission links and the J operation (by which 3rd, 5th and 7th bits of each word are inverted) to minimize decoder distortion.*)

Comments :

i) μ to A direction

There is no problem if the all-zero word is inhibited on 1544 kbit/s transmission links in the μ-law country.

ii) A to μ direction

One A-law character signal and one digital data signal are subjected to the following distortion :

- The A-law character signal "00101010" (decoder output value number in A-law = -128) is converted into the code "00000001" by the J operation and the Z operation at the MSC, and then into the μ-law character signal "00000001" (decoder output value number in μ-law = -126) by the J operation and the A → μ conversion at the gateway switch in the μ-law country (see Table 1).
- The data signal "00000000" in the A-law country is converted into "00000001" in the μ-law country.
- However, this distortion does not make the matter worse since the μ-law character signal "00000000" (decoder output value number in μ-law = -127) which corresponds to the A-law character signal "00101010" (decoder output value number in A-law = -128), and the digital data signal "00000000" would be inhibited on 1544 kbit/s transmission links in the μ-law country in any case.

In Figure 1, the other equipment related to interworking is also described. For instance, echo suppressors should be removed for voiceband data signals and digital data signals.

*) The desirability of incorporating the J-code operation should be studied further.

TABLE 1

Relation between the codes

μ-law code	A-law code	Code converted with J operation
(-127) 00000000	(-128) 00101010	00000000
(-126) 00000001	(-127) 00101011	00000001
:	:	:
(-85) 00101010	(-86) 00000000	00101010
(-84) 00101011	(-85) 00000001	00101011
:	:	:
(-3) 01111100	(-2) 01010100	01111110
(-2) 01111101		
(-1) 01111110		
(-0) 01111111	(-1) 01010101	01111111

( ) : Decoder output value number of each encoding law (see CCITT Recommendation G.711).

5. Data link for CCITT Signalling System No. 7

Since the transmission process of CCITT Signalling System No. 7 can avoid long zero strings with inversion of all bits, it was agreed in Working Party 2 of CCITT Study Group XI that for a 64 kbit/s signalling link in the μ-law country the inversion should be applied; however, in the A-law country it should not be applied. The process for interworking between the two countries was not clearly defined / 1 /

KDD proposes that the inversion should be applied in case of interworking between countries using different standards, and that the inverters should be placed at the gateway switches of both countries. If so arranged, the MSC need not recognize the signalling channel and this channel can be handled as an ordinary data channel. Also the all-zero word problem on 1544 kbit/s transmission links between the gateway switch in the μ-law country and the MSC is solved.

The configuration is also shown on Figure 1; particularly observe note b).

6. Conclusion

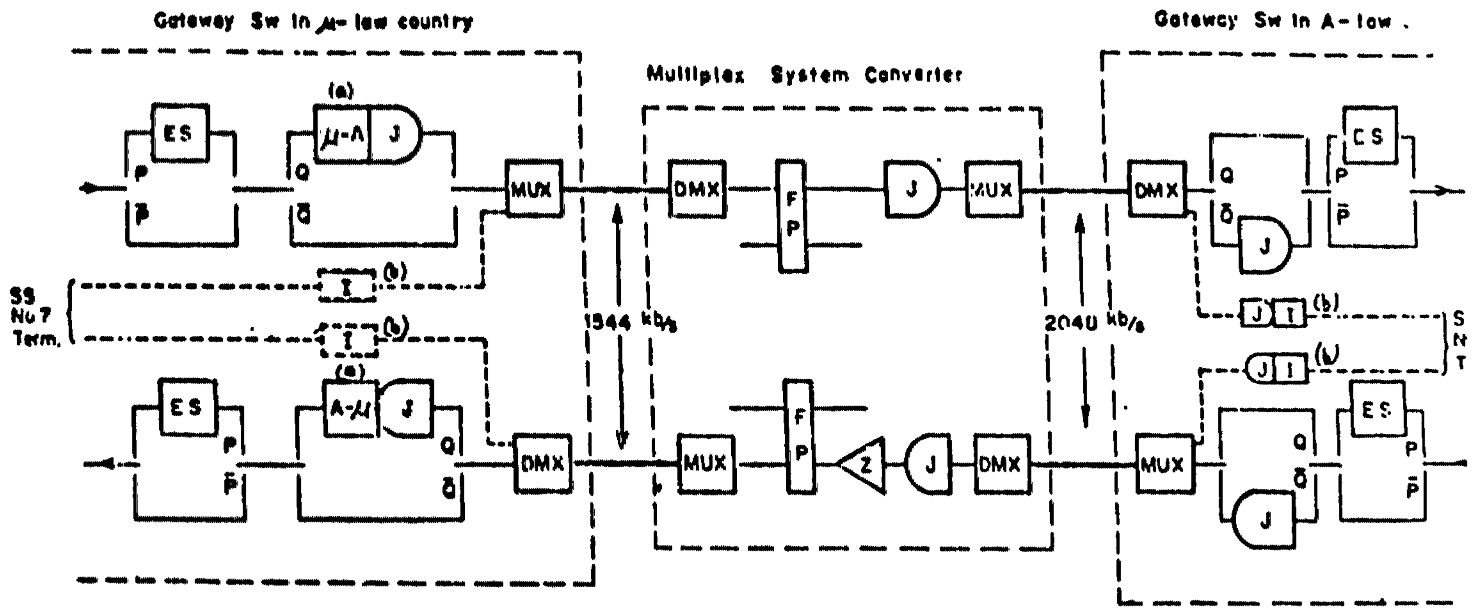
The interworking configuration described in Figure 1 is proposed to meet the interworking requirements. A method of interconnecting signalling links for CCITT Signalling System No. 7 is also proposed. In this proposal, the MSC need not identify whether voice or data is carried.

The insertion or removal of bit operations is required at the gateway switches of both sides according to the services. However, this kind of control is necessary anyhow, e.g. for echo suppressors.

The proposal, therefore, is considered to be worthy of further study as a possible solution to the problems of interworking between two standards.

BIBLIOGRAPHY

1. CCITT COM XVIII Study Period 1977-1980 — Contribution No. 245 (Working Party XI/2). (Report on the meeting held in Geneva from 22-26 January, Part V, Section 2.3)



Path Selection	
(1) Voice	: P , Q
(2) Voice-band data	: P̄ , Q
(3) Digitally generated data	: P̄ , Q̄

J : Inverts Bits 3,5,7 of each word

Z : Substitutes 00000001 for 00000000

I : Inverts all Bits of each word

ES : Echo suppressor

FP : Flexibility points (any location between DMX and MUX in MSC)

a) Recommendation G.711 will be amended to clarify that even-bit inversion is included in the A-μ and μ-A conversion process.

b) The I bit inversion in the CCITT Signalling System No. 7 path in the μ-law country may be considered integral to the Signalling System No. 7 terminal equipment. The I bit inversion in the same path in the A-law country may be provided by the gateway switch or by the Signalling System No. 7 terminal. Thereafter the Signalling System No. 7 path can be considered and treated as a digitally-generated data path.

Figure 1 - Interworking configuration

Annex 4

(to Question 14/XVIII)

Statement of Study Group VII as regards data problems involved with 64 kbit/s paths

During the last study period, Study Group VII has identified four different cases of interworking between data networks at a 64 kbit/s level :

- an X.50 or X.51 multiplexed stream;
- a 48 kbit/s data channel carried in a 64 kbit/s time slot (See Recommendation X.50 bis or X.51 bis);
- an X.75 digital link;
- an X.60 digital signalling link.

Conforming to Draft Recommendation G.72x, it might be necessary to add to existing X series Recommendations in the future. However, these additions can only be studied when interworking between PCM multiplexes based on different standards is defined.

Therefore Study Group VII asks Study Group XVIII to reply as soon as possible to Question 14/XVIII. In particular, a list of important features as regards digital data is given below :

- Will the 64 kbit/s telephone and data paths be shared or not ? If yes, on which basis ?
- Will signalling links (X.60, X.75, - -) be carried under specific paths or not ?
- Between two 64 kbit/s accesses, one located in an A-law country and the other in a µ-law country, will the 64 kbit/s bit stream be altered or not ?

QUESTION [illegible]/XVIII - Interfaces in digital networks

(continuation of Question 6/XVIII, studied in 1977-1980)

Considering

(a) That interface specifications are necessary to interconnect digital network components (line sections, multiplex equipment) to form an international digital path.

(b) That international digital paths can be interconnected through digital switching equipment and terminated in digital terminal equipment to form an international digital connection.

(c) That an international digital path and an international digital connection provide for the transmission of a digital signal (bits) at a specified hierarchical bit rate, independent of the service carried by the digital signal.

(d) That to ensure the interconnection of network components for the transmission of digital signals (bits) it is sufficient to specify physical, functional and electrical characteristics of the interface.

(e) That aspects of digital interfaces relating to the content (e.g., signalling protocols) of the digital signal transmitted over the interface are dealt with in other Questions of SGXVIII or by those study groups concerned with the service carried by the digital signal.

(f) That Recommendation G.703 specifies digital interfaces for interconnection of digital network components at hierarchical bit rates only.

(g) That interfaces at hierarchical bit rates for purposes other than directly providing transmission interconnection on an international digital connection (e.g., timing control distribution) may require specification.

(h) That Recommendation G.703 is referred to in other Recommendations on line sections and on terminal, multiplex and switching equipment.

(i) That interfaces at non-hierarchical bit rates shall be specified in the relevant equipment recommendations.

(j) That the evolution of digital technology may require the specification of hierarchical levels at bit rates other than those specified at present in Recommendation G.703.

(k) That for some interfaces of Recommendation G.703 the values for jitter require further study and for other interfaces the need for jitter specification and if appropriate the values for jitter have to be established.

1. Should interfaces other than those specified in Recommendation G.703 be recommended ?

2. What characteristics should be recommended for these interfaces, including:

   - Electrical characteristics
   - Functional characteristics
   - Physical characteristics
   - Any restrictions on the digital signals crossing the interface.

3. For interfaces presently quoted in Recommendation G.703:

   - Is there a need for additional specifications for jitter and wander ?

   - If so, what values should be specified ?

4. What is the impact that new transmission media (e.g. optical fibres) will have on interfaces ?

QUESTION 16/XVIII - Performance characteristics of PCM channels at audio frequencies

(Continuation of part of Question 8/XVIII, studied in 1977-1980)

Considering

- that some specification items in the Recommendations G.711 and G.712 need to be completed;

- that within a widespread digital network, it is envisaged that a telephony connection will ultimately require only a single encoding/decoding process for each direction of transmission;

- that independent encoder and decoder will be incorporated in each telephony connection and thus, separate transmit and receive side specifications at audio frequencies are needed;

- that for application in local area or with digital exchanges, provision for 2-wire analogue interface should also be considered;

a) What modifications to existing Recommendations G.711 and G.712 should be made ? For example, the value for longitudinal balance should be studied and specified. Also the necessity for the digital sequence for reference frequency and the necessity for high pass filtering in analogue to digital converters should be studied.

b) What are the values and limits to be specified for the audio frequency performance characteristics of PCM channels measured at the 2-wire point ?

c) What are the values and limits to be specified for the performance characteristics of PCM channels at audio frequencies when the transmit side and receive side are measured separately ?

Note 1 : The measuring method for longitudinal balance is under study in Study Group IV. Study Group XVI is also studying this matter.

Note 2 : With respect to the digital sequence for reference frequency, Recommendation G.101 should also be considered.

(22)

Annex 1

(to Question 16/XVIII)

Longitudinal balance specifications

The item contained in paragraph 4.3 of Recommendation G.712 (longitudinal balance) was discussed taking note of COM XVIII-No. 77 (FRG) and COM XVIII-No. 271 (Canada-BNR). These two Contributions are appended to this Annex.

Study Group XVIII identified the urgent need to determine the values for longitudinal balance as well as the corresponding measuring method. Although Study Group IV is aware that a figure for longitudinal balance can only be given if the corresponding measuring method is clearly defined, a specification for the measuring method does not exist as yet. For this reason, it is proposed that Study Group IV should pick up this matter during the next study period and prepare a Recommendation concerning the measuring method of longitudinal balance on equipment inputs and outputs.

In view of this situation, it seems not to be advisable to complete paragraph 4.3 of Recommendation G.712 still during the current study period.

Appendix 1

(to Annex 1 to Question 16/XVIII)

Amendment of Recommendation G.712 : Specification of impedance unbalance

(Contribution from the Federal Republic of Germany)

1. Background

Item 4.3 in Recommendation G.712 is designated since 1972 as being "under study". However, no contribution has since been presented. Nevertheless it is desirable to complete G.712 in this respect.

2. Proposal for item 4.3 in Recommendation G.712

"4.3 Impedance balance ratio

The impedance balance ratio, measured by means of the circuit defined in Recommendation O.121, Fig. 1, should not be less than 46 dB in the range 300 to 3400 Hz."

3. Remarks

3.1 Terminology

The term "longitudinal balance", so far employed in Recommendation G.712, is rarely used in CCITT publications. Instead, "impedance balance ratio" is used in Recommendation O.121. Other terms are employed elsewhere. An ad hoc Working Party of Study Group V and Study Group XVI (London, December 1975) has proposed a set of new terms in respect of unbalance (cf. Doc. COM XVI-No. 7) the discussion of which in Study Groups V and XVI is not concluded.

3.2 Practical usefulness

The method prescribed in Recommendation O.121 is well established and is used in existing commercial measuring equipment. The IEC recommends the same method for impedance unbalance measurements in the field of electro-acoustics (IEC Publ. 268-3).

3.3 Numerical value

With respect to possible cross 1k due to impedance unbalance, the values of unb..ance of connecting cables in a stacion are of more importance than those of the audio-frequency terminals of a PCM multiplex equipment. Accordingly, for the latter, a value of 46 dB, as stated in Recommendations K.10 and Q.45, will be sufficient. It does not seem necessary to relax the requirement in the range 300 to 600 Hz as in Recommendations K.10 and Q.45 since G.712 relates to four-wire ports where problems with feed coils do not arise.

Appendix 2

(to Annex 1 to Question 16/XVIII)

Proposal for longitudinal balance specifications for inclusion in Recommendation G.712

(Contribution from Canada : Bell Northern Research)

Abstract

This contribution proposes a set of values and the associated test m od for longitudinal balance for inclusion in Recommendation G.712.

1. Introduction

In the preliminary reply to Question 9/XVIII (COM XVIII-No.234, Period 1977-1980) Working Party XVIII/2 stressed the need to complete Recommendation G.712 during the current study period. One of the items presently under study in Recommendation G.712 is longitudinal balance.

Measurements of longitudinal balance, in particular, depend on the test method used. This ccatribution therefore proposes a set of values and the associated test method for longitudinal balance for inclusion in Recommendation G.712.

2. Test Method

In the past, several test methods for longitudinal balance measurements have been used in Canada and elsewhere in North America.

In an attempt to reach agreement on a single method of testing longitudinal balance "IEEE STANDARD 455-1976" was developed. This standard describes the test procedure for measuring longitudinal balance of telephone equipment operating in the voice band and is gaining wide acceptance in Canada. Comparative tests of longitudinal balance on various devices with four test sets constructed independently according to the IEEE Standard demonstrated the reproducibility of measurements and were submitted to CCITT (COM XVI-No.73, Bell-Northern Research Period 1973-1976). CCITT Study Group V is also considering at present the merits of the IEEE Standard (COM V-No.22, COM XVI-No.43, Period 1977-1980).

Figure 1 shows the test circuits for measuring longitudinal balance according to the IEEE standard. The degree of longitudinal balance- the ratio of the disturbing longitudinal voltage Vs and the resulting metallic voltage Vm of the network under test, expressed in dB,- is:

$$\text{Longitudinal-balance} = 20 \log_{10}\left|\frac{V_s}{V_m}\right| \text{ [dB]}$$

We propose to use the test method described in "IEEE STANDARD 455-1976" and as shown in Figure 1 when measuring longitudinal balance of PCM multiplex equipment.

3. Longitudinal Balance Requirements

Figure 2 shows one test method which has been widely used in Canada and elsewhere in North America in the past. Other test methods were also used.

Longitudinal balance for the test method shown in Figure 2 was expressed as

$$\text{Longitudinal balance} = 20 \log_{10}\left|\frac{V_s}{V_m}\right| \text{ [dB] and}$$

minimum longitudinal balance requirements for the 4-wire ports of PCM multiplex equipment were:

200 Hz	86 dB
1000 Hz	80 dB
3000 Hz	78 dB

PCM multiplex equipment designed and manufactured meeting these requirements is operating satisfactorily in the network today.

Taking these existing requirements as a base, conversion factors had to be derived to determine equivalent longitudinal balance requirements for use with the proposed test method of Figure 1.

Theoretical analysis and measurements conducted at Bell-Northern Research show that for all practical purposes a conversion factor of 0 dB can be used over the voice frequency band as long as the longitudinal impedance of the test specimen exceeds 30 k ohms. Since 4-W ports of PCM multiplex equipment generally have a longitudinal impedance in the order of 100 k ohms a 0 dB conversion factor can be used.

Therefore, we propose to include the following longitudinal balance requirements for the 4-W ports into Recommendation G.712:

Frequency [Hz]	Minimum Longitudinal Balance [dB]
200	86 dB
1000	80 dB
3000	78 dB

4. Proposal for Inclusion in Recommendation G.712

Summarizing the considerations under Items 1, 2 and 3, we propose to amend Recommendation G.712 as follows:

4.1 Rec.G.712, Item 1, Third Paragraph

Amend the third paragraph to read:

'The values and limits specified are those which should be obtained in 4-wire measurements using two PCM multiplex terminal equipments connected back-to-back (except for 5.3 below) and with the input and output ports of the channels terminated with their nominal impedance (except for 4.3 below).'

4.2 Rec.G.712, Item 4.3

Delete 'Under study'. Insert the following text:

Longitudinal balance should be measured in accordance with the driving test circuit and the terminating test circuit shown in Figure 1 of Annex 2

Longitudinal balance is defined as

$$20 \log_{10} \left| \frac{V_s}{V_m} \right| [\text{dB}]$$

The minimum longitudinal balance should be:

200 Hz	86 dB
1000 Hz	80 dB
3000 Hz	78 dB

4.3 Rec.G.712, New Annex 2

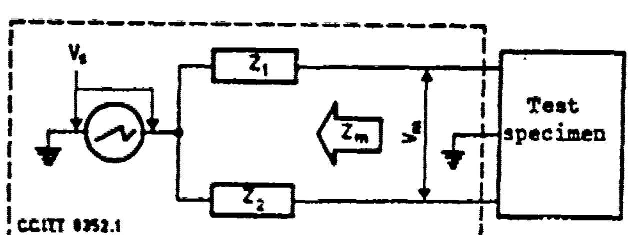
ANNEX 2
(to Recommendation G.712)

Test Circuit for Longitudinal Balance Measurements

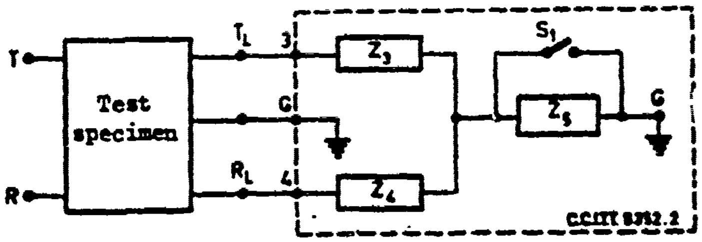
Figure 1 shows the standard driving and standard terminating test circuits for longitudinal balance measurements. Nominal impedance values are:

Z_1, Z_2, Z_3, Z_4	368 ohms
Z_5	2000 ohms
Z_m	736 ohms

A full description of the test procedure for measuring longitudinal balance of telephone equipment operating in the voice band is given in 'IEEE STANDARD 455-1976'.

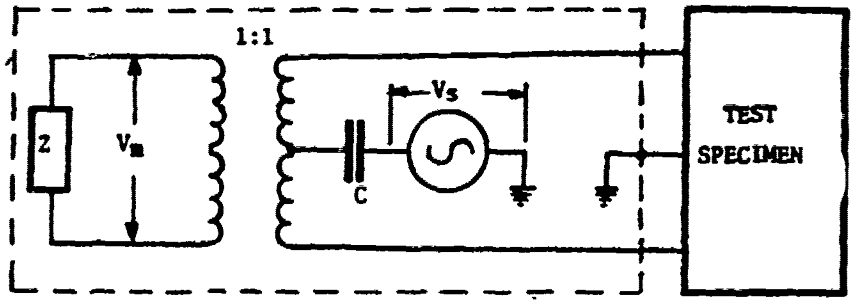


(a) Standard driving test circuit for measurement of single port and two port networks.



(b) Standard terminating test circuit for measurement of two port networks.

Figure 1 - IEEE test method for longitudinal balance measurements



Z = 600 ohms
C = 0.5 µF

Figure 2 - Test method for longitudinal balance measurements used in the past in Canada

Annex 2

(to Question 16/XVIII)

Study Group XV (Geneva Meeting, 25 June-13 July 1979) extract from the Report of the Working Party on Echo Suppressors (Contribution COM XV-No. 324)

"The Echo Suppressor Working Party took note of the extract from the preliminary report to Question 17/XVIII and of the extract of preliminary reply to Question 1/XVIII and the need for disabling of an echo suppressor or canceller when used in an integrated services digital network. Present designs of echo suppressors and cancellers include an external enable/disable control but do not presently recognize a signal which indicates bit integraty is required. Further information will be required by Study Group XV before such disablers can be designed.

Study Group XVIII is further advised that Recommendation G.712 (Figure 1/G.712) does note require the use of high pass filtering in A/D conventers how frequency interference from power supplies is therefore not attenuated. This makes it necessary for any following digital equipment (particularly those using speech detectors) to provide high pass filtering. Study Group XVIII are asked to consider the possibility of providing appropriate high pass filtering in A/D conventers recommended in G.712 having due regard to the total system economics".

Annex 3

(to Question 16/XVIII)

A proposal for specifications on performance characteristics of 2-wire PCM channels at audio frequencies

(Contribution from Nippon Telegraph and Telephone Public Corporation)

1. Introduction

CCITT Rec. G.712 specifies performance characteristics only for 4-wire PCM channels at audio frequencies. It should be noted, however, that PCM multiplex equipments with 2-wire PCM channels have been widely used for transmission between 2-wire analog exchanges, or for pair gain systems in local areas, and will be used for digital local exchanges. CCITT Study Group XI is preparing the draft recommendations for digital transit and local exchanges , where 2-wire analog interfaces as well as 4-wire analog interfaces are involved.

NTT is of the opinion that Study Group XVIII should study the performance characteristics of 2-wire PCM channels at audio frequencies, and specify the values and limits as soon as possible in order to avoid the diversification of the specification for 2-wire PCM multiplex equipments.

This contribution proposes performance characteristics of 2-wire PCM channels as a base for the study in the 1981-1984 study period.

2. Proposed specifications for 2-wire PCM channels at audio frequencies
A transmission path A-B shown in FIGURE 1 is defined as a 2-wire PCM channel. The values and limits to be specified are those which should be obtained in 2-wire measurements using two PCM multiplex equipments with 2-wire PCM channels back-to-back and with input and output ports of the channels terminated with their nominal impedance.

Further study is required for the separate specifications of the send and the receive sides of 2-wire PCM channels.

2.1 Attenuation/frequency distortion (Corresponding to Sec.2/G.712)
The variations with frequency of attenuation of any channel should lie within the limits shown in the mask of FIGURE 2.

The reference frequency is 800 Hz. The input power level should be 0 dBm0.

2.2 Return loss (Corresponding to Sec.4.2/G.712)
The departure from the nominal value, measured as return loss against the nominal value, should not be less than 12 dB over the frequency range 300 to 600 Hz and less than 15 dB over 600 to 3400 Hz.

2.3 Longitudinal balance (Corresponding to Sec.4.3/G.712)
Further study is needed.

Note: Longitudinal balance for 2-wire PCM channels must be specified. Rec.Q.45 may be referred to.

2.4 Stability and echo loss (New items for 2-wire PCM channels)
Since a 2-wire PCM multiplex equipment contains hybrid circuits for conversion between 2-wire and 4-wire, the transmission loss of the path T-A-U shown in FIGURE 1 from the point of view of stability and of echo should be specified.

It should be noted, however, that the loss of the path T-A-U depends on the losses of the pads, X and Y and the characteristics of the bandpass filters shown in FIGURE 1. Since the values of the losses due to these circuits are left to Administrations involved, the balance return loss component of the total loss for the path T-A-U, defined in Rec. G.122 may be a possible specification for the stability and the echo requirements.

The measuring method as well as the values for the balance return loss and the echo balance return loss should be further studied and specified.

2.5 Discrimination against out-of-band input signals (Corresponding to Sec.6 /G.712)
Rec. G.712 Sec.6 specifies the loss in the range 4.6-72 kHz. For 2-wire PCM channels, it might be necessary to specify the loss arround 50 Hz in order to discriminate the interference from power cables. Further study is required.

2.6 Go-to-return crosstalk (Corresponding to Sec.13/G.712)
Since it is difficult to measure go-to-return crosstalk for 2-wire PCM channels, this item is left unspecified.

2.7 Other items
For items other than those presented above, the same specifications as those recommended in Rec. G.712 for 4-wire PCM channels should basically be applied to 2-wire PCM channels.

3. Conclusion
NTT proposes that Rec. G.712 should involve performance characteristics of not only 4-wire PCM channels, but also 2-wire PCM channels. The values and limits presented above are proposed as a base for further study.

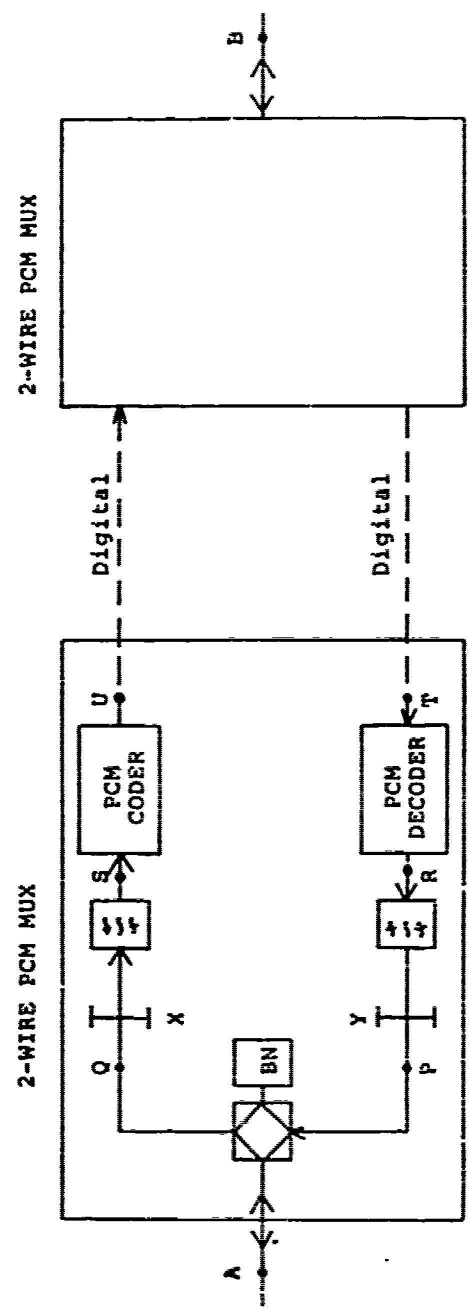


Figure 1 - 2-Wire PCM channel

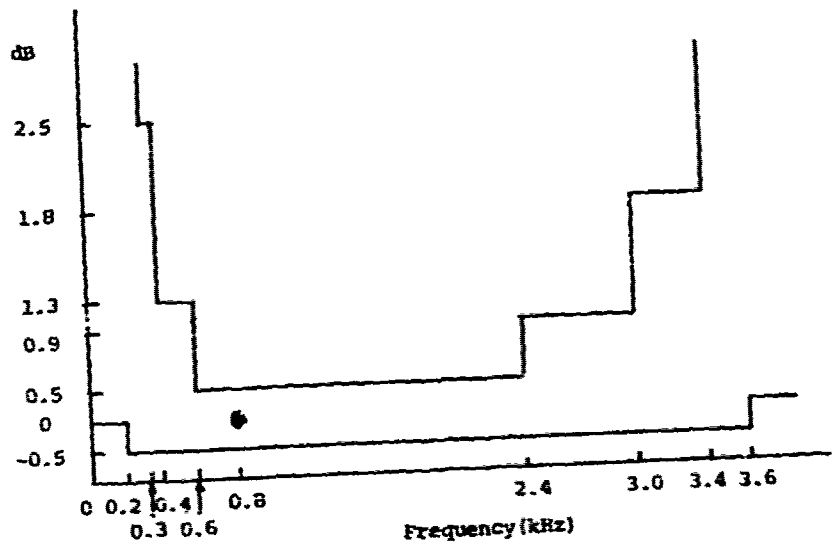


Figure 2 – Attenuation/frequency distortion for 2-wire PCM channels

Annex 4

(to Question 16/XVIII)

Proposed principles and tentative values for parameters of separate specification of PCM channel at audio frequencies

1. Should the requirements of G.712 be relaxed in any case ?

Considerable discussions took place on this question of whether, in defining the separate limits for the send and receive sides, it would be acceptable to have a situation whereby a most adverse combination of send and receive sides would have an overall performance slightly inferior to G.712. The general feeling was that for most parameters the separate limits should be defined in such a way that G.712 is always met. However, some delegates were of the opinion that for some parameters at least (e.g. total noise, crosstalk, gain versus level) it would be unrealistic to set such separate limits and they would be willing to accept that for a most adverse combination, which would occur only very infrequently, then an overall performance very slightly inferior to G.712 would be acceptable. This point could not be resolved and it was agreed to consider the question for each parameter later.

2. Should separate limits be established for every parameter in G.712 ?

All the parameters in G.712 were considered and Table 1 is a summary of the agreement reached on how each parameter should be dealt with.

3. Should there be any difference in the limits applicable to A-law or µ-law systems ?

Everybody agreed on the desirability of having common limits but some delegates expressed the opinion that for certain parameters the differences between the theoretical performance and the limits given in G.712 are rather small and it might be necessary, in certain cases, to define different limits. This point will have to be resolved when individual parameters are discussed.

4. What basis of allocation of limits should be defined ?

It was agreed that no single principle could be adopted but rather the allocation would be dependent upon the parameter in question. In some cases it was recognized that voltage or power summation would apply and furthermore it would not always be appropriate to allocate limits equally for the send and receive sides.

5. Should a margin be included for measurement error ?

At first confusion arose because of an apparent discrepancy between the English and French versions of the second sub-paragraph of paragraph 1 of G.712. It was agreed that the spirit of the English version should be followed and that the limits to be defined should not include a margin to take account of the measurement inaccuracy of test equipment. However, performance limits should be met in all cases making due allowance for any inaccuracy in the testing techniques.

The correct French translation of the English version is as follows :

"Les limites de qualité - - - - tous les cas, sauf en cas d'imprécision éventuelle des méthodes de mesure appliquées."

6. Tentative values for parameters

On the basis that the limits would be open to future amendment, it was agreed that some tentative values should be proposed for some of the parameters that require separate specification. It was felt by delegates that if numbers exist then this will encourage people to carry out measurements and further studies in order to determine whether the limits are feasible and realistic.

In proposing values, the basic concepts of a standard send side and a standard receive analyser were accepted. The definitions of such hypothetical devices are as follows :

a) A standard send side is a hypothetical device which is absolutely ideal. i.e., a perfect analogue digital converter preceded by an ideal low pass filter (assumed to have no frequency attenuation distortion and no envelope delay distortion) or it is a digital processor which simulates the above.

b) A standard receive analyser is a hypothetical device which is either a standard receive channel that is absolutely ideal, ie a perfect digital to analogue converter followed by an ideal low pass filter (assumed to have no frequency attenuation distortion and no envelope delay distortion) on it is a digital processor which simulates the above.

In practice it is envisaged that test equipment based on these concepts, will become available. Although such equipments might not be perfect they should have adequate accuracy.

i) Attenuation - frequency distortion

Each of the limits for the send and receive sides should be half of the G.712 limit.

Comment : this reflects the agreement already reached at the last meeting of Working Party XVIII/2.

ii) Envelope delay distortion

Each of the limits for the send and receive sides should be half of the G.712 limit. In addition, the upper limits for the value of the minimum group propagation delay should be half of the G.712 limit.

Comment : some delegates expressed doubt about the feasibility of carrying out such measurements in practice but at least this requirement should be considered as a design objective. One delegate suggested that the presence of a hum rejection filter only on the send side of some equipment might mean that more than half of the overall limit should be allocated to the send side.

With regard to the limits for the minimum group propagation delay, some delegates considered that a slightly larger allowance should apply to the encoding function because, for example, it would seem reasonable for a single channel encoder to take up to 125 μs to produce an output code word. In the decoder, the reconstructed output is available almost immediately after the application of the input code word. Even with 100 % sample and hold the consequential delay is only 62.5 μs. It might be appropriate to allocate the separate limits by applying the following equation :

$$2F + 125 + 62.5 = \text{G.712 limit}$$

where F is the delay of the filters.

iii) Adjustment of relationship between encoding law and audio level

The gain accuracy of each of the send and receive sides should be ± 0.3 dB.

Comment : this is the same requirement that already exists in G.712 paragraph 17.

iv) Short and long term stability

Each of the send and receive limits should be half of the G.712 limits

Comment : none.

v) Total distortion including quantizing distortion

Method 1 - Noise

The following limits are based upon calculations of what separate limits are required to guarantee the G.712 overall performance for any combination of send and receive sides. See the commentary.

TABLE 2

Signal to total distortion as a function of input level (dB) - Method 1

Input level dBm0	A-law		μ-law	
	Send side	Receive side	Send side	Receive side
-3	27.9	29.1	28.1	28.9
-6	35.5	36.7	35.7	36.5
-27	35.1	36.4	34.9	36.0
-34	33.4	34.7	33.2	34.3
-40	28.8	30.1	30.0	30.5
-55	13.7	15.0	15.0	15.5

Note : The mask is constructed by interconnecting the points by straight lines.

Method 2 - sine wave

The following limits are based upon calculations of what separate limits are required to guarantee the G.712 overall performance for any combination of send and receive sides. See the commentary.

TABLE 3

Signal to total distortion as a function of input level (dB) - Method 2

Input level (dBm0)	A-law				μ-law	
	No. 252		Delayed BI		Delayed BI	
	Send side	Receive side	Send side	Receive side	Send side	Receive side
0	34.5	36.0	35.3	36.4	35.3	36.4
-30	34.5	36.0	35.3	36.4	35.3	36.4
-40	28.5	30.0	29.3	30.4	29.3	30.4
-45	23.5	25.0	24.3	25.4	24.3	25.4

Note : The mask is constructed by interconnecting the points by straight lines.

Comment : the limits above in Tables 2 and 3 are extracted from the Delayed Contribution BI except for some of the values shown in Table 3 which are extracted from Contribution COM XVIII-No. 252.

They are all based on calculations to determine what separate send and receive performances are required in order to guarantee that for any combination of send and receive sides, the existing overall limits contained in G.712 are always met for any input level. The overall limit is apportioned on a 2 : 1 basis in favour of the send side. The limits so derived leave little manufacturing margin and all delegates agreed that these values might not be achievable. However, it was agreed that they should form the basis for further study.

The Annex of Delayed Contribution BI is a theoretical analysis of the above and it is reproduced as Appendix 1 to this Report.

For the theoretical background to the Italian document reference can be made to :-

ALTA FREQUENZA No. 2 Volume XLIV - 1975, pages 57-65, titled

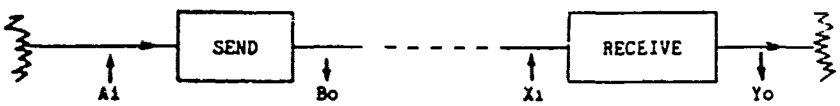
"Accuracy assignment and separate performance measurement in the transmitting and receive ends of PCM multiplexers"

by G.G. Capecchiacci and A.M. Molinari.

These two references should be studied to decide which assumptions are more valid because there are some differences which could not be resolved.

vi) Idle channel noise

Figure 1 shows the measuring arrangements together with a summary of the various proposals. In comparing the proposals for the send sides, the only difference is the limits. For the receive side, some delegates considered it necessary to introduce the signal shown as Xi which, in effect, is a test to ensure that the decoder quantum step sizes are not excessively large. Other delegates considered that this extra complexity was not necessary since other requirements, such as quantizing distortion and linearity, ensure that the quantum step sizes are reasonably precise. This point requires further study.



	174 (UKPO) 252 (Italy) 284 (France) 294 (FRG)	BI (NTT)	306 (ATT)
Ai	-	-	-
Bo	- 66 dBm0p	- 67 dBm0p	- 68 dBm0p
Xi	-	- 67 dBm0p coded white noise	i) idle code ii) Φ
Yo	- 75 dBm0p	- 65 dBm0p	i) - 75 dBm0p ii) - 65 dBm0p

Φ - 68 dBm0p coded white noise with variable dc bias of ± 7 quantum steps

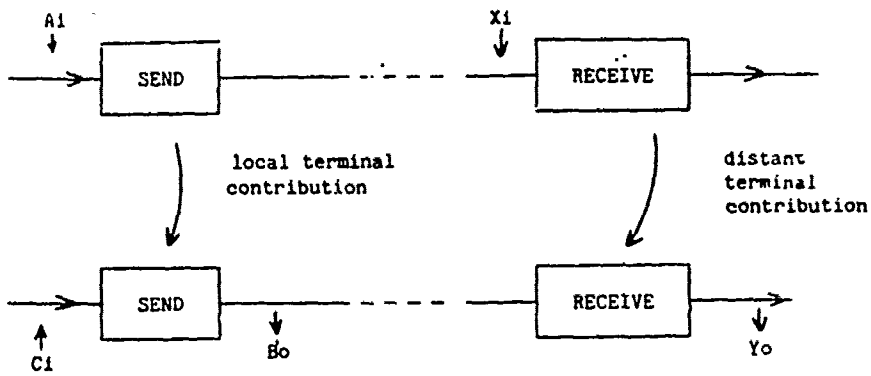
Figure 1 - Idle channel noise measurement arrangements and limits

vii) Inter-channel crosstalk

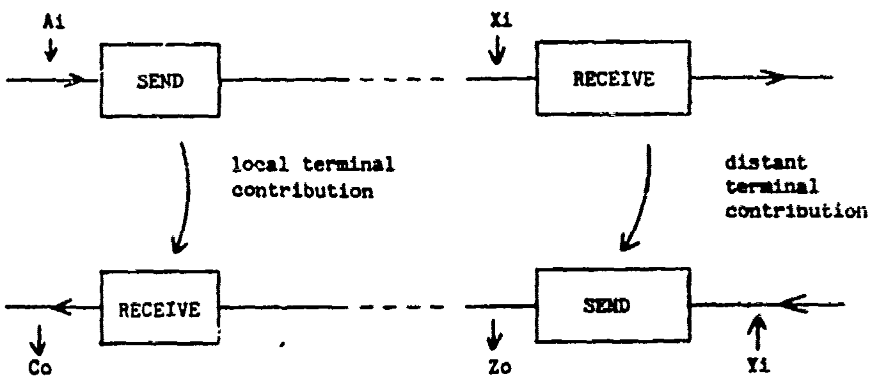
For both far end and near end crosstalk, it was recognised that there were two contributions to each. These are termed local and distant terminal contributions.

After considering the various contributions and after considerable discussion, it was agreed that the method of defining crosstalk in G.712 is not entirely satisfactory since the gain enhancement effects that can occur in encoders at very low input levels mask the real crosstalk. A test method that effectively evaluates the analogue crosstalk is much more appropriate and it was noted that a number of earlier contributions have been made on this subject (COM SpD-No. 59, August 1970 and COM XVIII-No. 8). They are all based upon the concept of adding a low level activating signal into the disturbed channel.

Figure 2 illustrates the measuring arrangements that are appropriate as a basis for further study.



a) Far end crosstalk (FEXT)



b) Near end crosstalk (NEXT)

Figure 2 - Near and far end crosstalk

Key :

Ai	disturbing signal	Xi	disturbing signal
Bo	local terminal contribution to FEXT	Yo	distant terminal contribution to FEXT
Ci	activating signal	Yi	activating signal
Co	local terminal contribution to NEXT	Zo	distant terminal contribution to NEXT

viii) Go-to-return crosstalk

The same comments apply to this parameter as stated above for near end crosstalk. However, the eventual limits may be somewhat different because of the less stringent requirement of G.712 in respect of this parameter.

ix) <u>Variation of gain with input level</u>

It was noted that this was another parameter for which the two possible test methods represent a different level of stringency; the method based upon the use of sine wave signals being generally much more easy to satisfy. For the latter, a split of the overall limit on a 1 : 1 basis between the send and receive sides was not considered to pose any real difficulties.

The limits shown in Table 4 are based upon the figures in Delayed Contribution BI which are calculated assuming worst case addition for any combination of send and receive sides. The group considered them suitable as the basis for further study.

TABLE 4

Variation of gain with input level - Method 2 (sine)

Input level dBm0	Send or receive side limit (dB)
+ 3 ~ - 40	± 0.25
- 40 ~ - 50	± 0.7
- 50 ~ - 55	± 1.8

In the case of Method 1 (noise), it was not possible to agree on what basis the overall limit should be split. Some delegates favoured an equal split while others thought a split in favour of the encoder to be more appropriate.

All delegates agreed with the view that the requirement for the input signal range - 55 to - 60 dBm0 was very stringent and possibly unnecessary. Bearing in mind that these signal levels are down in the same order as crosstalk and noise signals present on the system, this point should be studied further. As a basis for this work, it was recognised that the Appendix of COM XVIII-No. 252 (Italy) is a useful document and this is appended to this report.

7. Additional observations

During the discussions a number of observations of a general nature were made. These points were not considered to be within the terms of reference of the group, but are nevertheless brought to the attention of Working Party 2/XVIII.

i) Some delegates noted the poor correspondence between the two methods recommended for testing total noise and linearity as defined in G.712. In particular, the linearity required at very low levels for the noise test method is very stringent and it is recommended that the need for this level of performance should be reassessed.

ii) Some delegates noted that some [illegible] of G.712 may be required to reflect the results of the study of separate specification of PCM channels.

TABLE 1

Parameter	New limits required for		Comment
	Send side	Receive side	
Attenuation frequency	✓	✓	
Envelope delay distortion and minimum group propagation delay	✓	✓	
Impedence and return loss	-	-	already covered in G.712
Longitudinal balance	-	-	
Idle channel noise	✓	✓	
Single frequency noise	-	-	already covered in in G.712. These are only applicable to receive side performance
Receiving equipment noise	-	-	
Discrimination against out-of-band input signals	-	-	already covered G.712. Only applicable to send side performance
Spurious out-of-band signals at output	-	-	already covered in G.712. Only applicable to the receive side performance
Intermodulation	-	-	Working Party 2 has previously agreed that it is not necessary to separately specify
Total distortion including QD	✓	✓	
Spurious in-band signals at channel output	-	-	the Group considered that it was not necessary to separately specify this parameter

TABLE 1 (cont.)

Parameter	New limits required for		Comment
	Send side	Receive side	
Variation of gain with input level	✓	✓	
Interchannel crosstalk	✓	✓	both near end and far end crosstalk to be considered
Go to return crosstalk	✓	✓	local and distant terminal contributions to be considered
Interference from signalling	-	-	the Group considered that it was not necessary to separately specify
Relative levels at input and output	-	-	already covered in G.712
Short term and long term stability	✓	✓	
Adjustment of relationship between encoding law and audio level	✓	✓	this item is already effectively covered in G.712

Note : For some parameters the appropriate limits already appear in Recommendation G.712. However, they will need to be considered in terms of the measurements arrangement to be adopted.

Appendix 1

(to Annex 4 to Question 16/XVIII)

Separate specification on total distortion including quantizing distortion

(Contribution from Nippon Telegraph and Telephone Public Corporation)

1. Allocation of total distortion to sending and receiving sides

Notations used in the Appendix are shown in Figure 6. The following assumptions are considered here.

(1) In Figure 1, degradation in sending side, N_S, and degradation in receiving side N_R, are assumed to be added on a power-sum basis.

(2) In Figure 1, N_S is permitted α times as much as N_R. From circuit experiences, $\alpha = 2$ seems reasonable.

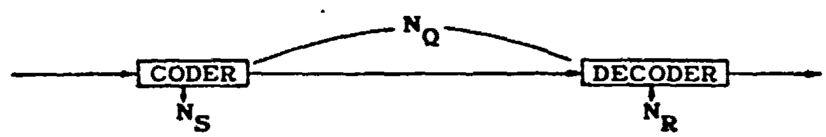


Figure 1 - Degradation in a codec

The signal-to-total distortion ratio for the end-to-end of the codec is written as

$$(S/N)_{SR} = S/(N_Q + N_S + N_R) \quad \ldots\ldots\ldots\ldots\ldots (1)$$

Measuring separately, S/N values for a coder and a decoder are respectively given as follows :

$$\left.\begin{aligned}(S/N)_S &= S/(N_Q + N_S)\\ (S/N)_R &= S/(N_Q + N_R)\end{aligned}\right\} \quad \ldots\ldots\ldots\ldots\ldots\ldots (2)$$

Now, from Eq. (1) and Eq. (2), the following inequalities can be written.

$$(S/N)_S = S/(N_Q + N_S) \geq 10^{\frac{x}{10}} \quad \ldots\ldots\ldots\ldots\ldots (3)$$

$$(S/N)_R = S/(N_Q + N_R) \geq 10^{\frac{y}{10}} \quad \ldots\ldots\ldots\ldots\ldots (4)$$

$$(S/N)_{SR} = S/(N_Q + N_S + N_R) \geq 10^{\frac{f}{10}} \quad \ldots\ldots\ldots (5)$$

When a coder satisfying Eq. (3) and a decoder satisfying Eq. (4) are connected, the end-to-end performance of the codec should satisfy Eq. (5). Then, the following condition should be met.

$$10^{-\frac{x}{10}} + 10^{-\frac{y}{10}} \leqq 10^{-\frac{f}{10}} + 10^{-\frac{g}{10}} \quad \ldots\ldots\ldots\ldots\ldots (6)$$

From assumption (2) mentioned above,

$$10^{-\frac{x}{10}} - 10^{-\frac{g}{10}} = \alpha\left(10^{-\frac{y}{10}} + 10^{-\frac{g}{10}}\right) \quad \ldots\ldots\ldots (7)$$

is obtained.

Now, curves can be drawn which represent Eq. (6) and Eq. (7) for given f and g. Since f and g are dependent upon the input level, ℓ , a number of curves can be obtained corresponding to ℓ . In order to guarantee the existing G.712 S/N limit when a coder and a decoder are interconnected, the level, ℓ_o, which maximizes (g - f), should be used. In Fig. 7(a), shaded area shows the region where the S/N of a coder-decoder pair is guaranteed to satisfy the existing G.712 S/N limit even in the worst case. The point P on the curve corresponding to Eq. (7) provides the desirable separated values which maximize the permissible margin from the theoretical S/N_Q.

In the contribution, COM SGXVIII - NO. 174, the point Q was selected for the allocation. Although the point Q can provide the larger margin, the end-to-end S/N performance cannot satisfy the G.712 S/N limit in the worst case at any input level except the level to minimize (g - f).

The allocation method described above enables to easily obtain the required S/N increase from the G.712 S/N limit and the margin for manufacturing.

2. Numerical examples for the sinusoidal measurement (Method 2)

Fig. 7(b) shows the separate specification for 8-bit μ-law encoding obtained with the S/N allocation method illustrated in Fig. 7(a). The required S/N increase is 2.3 dB for a sending equipment and 3.4 dB for a receiving equipment. The minimum margin, m_S and m_R, is 2.7 dB and 1.6 dB, respectively, which seems realizable from a manufacturing point of view. In this figure, the end-to-end S/N performance in the worst case is calculated to illustrate that any combinations of sending and receiving sides can meet G.712 S/N limit at any input level.

In Fig. 7(c), the separate S/N limits obtained above are applied to A-law encoding. The minimum margin, m_S and m_R, is 4.5 dB and 2.5 dB, respectively. The end-to-end S/N performance can satisfy G.712 S/N limit in the worst case.

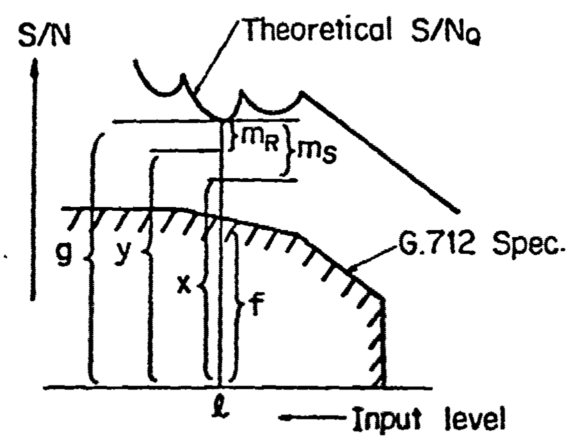
Accordingly, this separate specification may be regarded as a possible separate standards for Method 2 common to both μ-law and A-law encoding.

3. Numerical examples for the white noise measurement (Method 1)

When the cross point between Eq. (6) at the input level to maximize (g - f) and Eq. (7) at the input level to minimize (g - f) is selected as a separation point, the resultant minimum margin, Min. (m_S) and Min. (m_R), is too small to realize practical circuits in case of Method 1. It is, therefore, necessary to change the values of the S/N increase from the G.712 limit, depending upon the several input levels corresponding to the edges of the G.712 S/N mask (Figure 4/G.712). Fig. 8(a) shows a separate specification of 8-bit μ-law encoding for white noise measurement. The end-to-end S/N performance in the worst case satisfies the G.712 S/N limit at any input level. Minimum margin, Min. (m_S) and Min. (m_R), is 2.5 dB and 1.4 dB, respectively.

For A-law encoding, the separate S/N limits can be specified in a similar manner. Fig. 8(b) is an example of separate S/N limits for 8-bit A-law encoding. The separate specification guarantees to meet the G.712 end-to-end S/N limit in the worst case at any input level. However, from a manufacturing point of view, it seems extremely severe to clear this separate specification at low input levels where minimum margin, Min. (m_S) and Min. (m_R), is approximately 3 dB and 2 dB, respectively. Some relaxation may be necessary in this case.

According to the numerical examples shown above, it may be impossible to provide a common separate S/N limit for both μ-law and A-law encoding, when the end-to-end S/N performance for any combinations between sending side and receiving side is required to satisfy the G.712 S/N limit (Figure 4/G.712) at any input level. Further study may be necessary in this connection.



g : Theoretical S/NQ [dB]

f : G712 Specification [dB]

x : Sending side specification [dB]

y : Receiving side specification [dB]

m_S : Margin for manufacturing sending equipment [dB]

m_R : Margin for manufacturing receiving equipment [dB]

ℓ : Input level [dBm0]

Figure 6 - Explanation of the notation

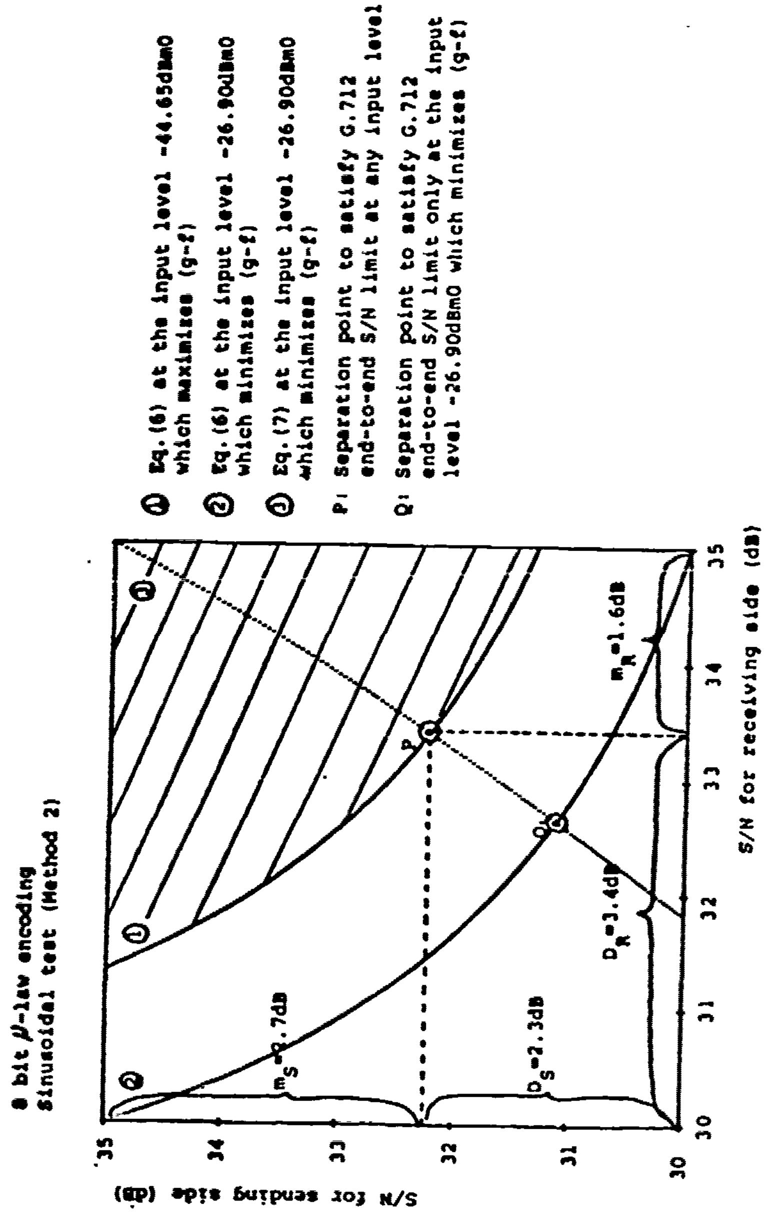


Figure 7(a) Separation method for total distortion

8 bit μ-law encoding
Sinusoidal test (Method 2)

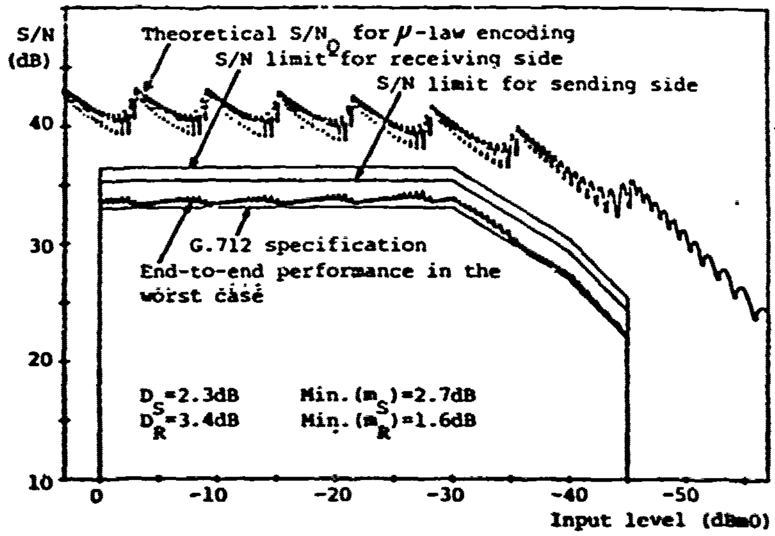


Figure 7(b) Separate specification applied to µ-law encoding

8 bit A-law encoding
Sinusoidal test (Method 2)

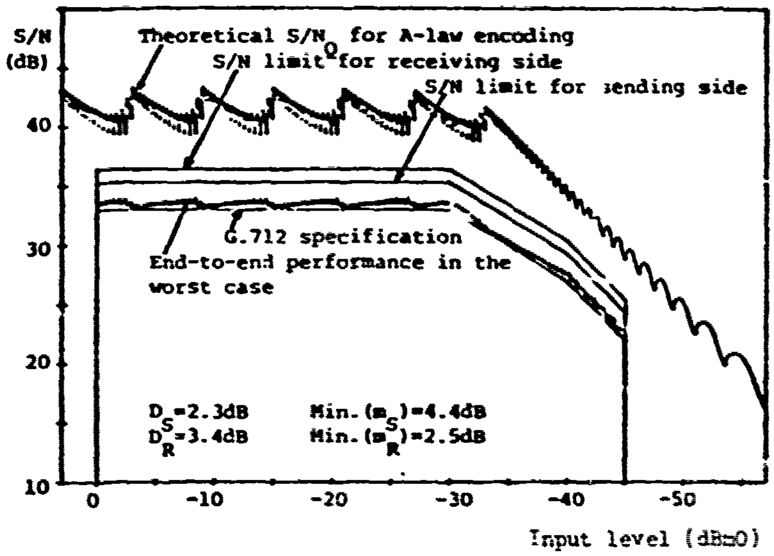


Figure 7(c) Separate specification applied to A-law encoding

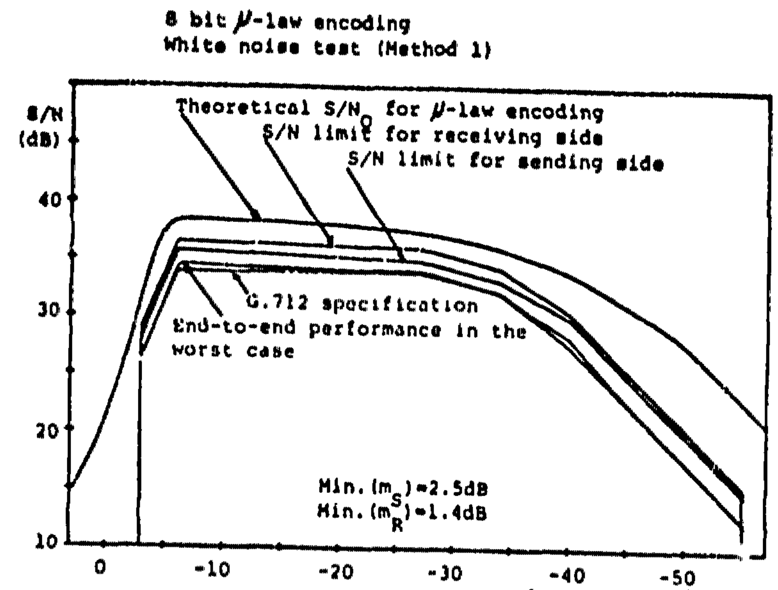
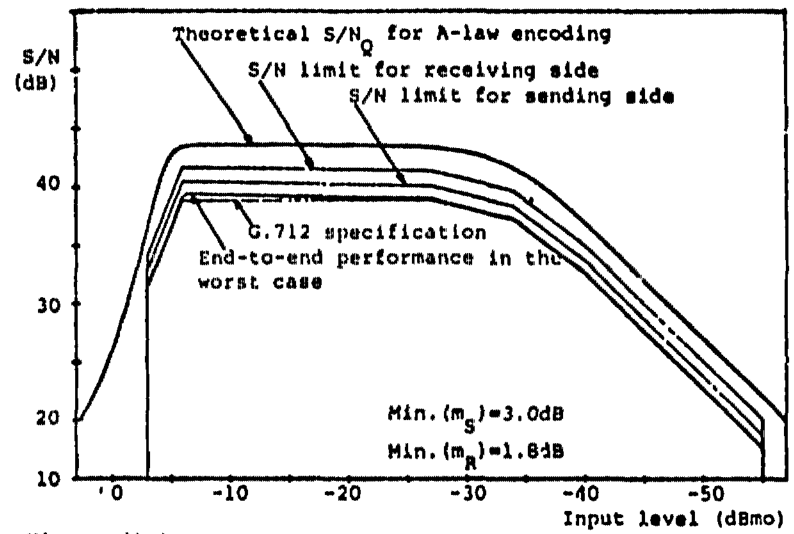


Figure 8(a) Separate specification applied to μ-law encoding

Input level (dBm0)	D_S (dB)	D_R (dB)
-3	1.8	2.6
-6	"	"
-27	1.0	2.1
-34	"	"
-40	2.4	2.9
-55	"	"

8 bit A-law encoding
White noise test (Method 1)



Input level (dBm0)	D_S (dB)	D_R (dB)
-3	1.6	2.8
-6	"	"
-27	1.2	2.5
-34	"	"
-40	"	"
-55	1.1	2.4

Figure 4(b) Separate specification applied to A-law encoding

Appendix 2

(to Annex 4 to Question 16/XVII.)

Variation of gain versus input level (noise signal, Method 1)

(Contribution from the Italian Administration)

Particular care has been taken when analyzing this performance because the relevant requirement of Recommendation G.712 seems to be very stringent.

Measurements on equipments of different manufacturers indicate that Recommendation G.712 is complied with good margins down to input signal level of -50 dBm0. Conversely the margins are reduced when the input signal levels are less than -50 dBm0. It has also been observed that Recommendation G.712, concerning gain versus level, with sine-wave signal (method 2) is less restrictive than Recommendation G.712 with noise signal (method 1); in fact the following table holds true :

Level (dBm0)	Margin with sine-wave signal	Margin with noise signal
-60	(not specified)	1 dB
-55	> 2 dB	0.5 dB
-50	> 0.7 dB	0.5 dB

Margin = Recommendation G.712 mask minus theoretical gain variation.

On the basis of these considerations a computer simulation has been carried out in order to reach "reasonable" separate limits valid for the send side and the receive side, respectively.

It will be possible to deduce from the following discussion that the simple division by two of the present Recommendation G.712 leads to an excessive requirement on the accuracy of the decision and reconstruction levels.

At signal levels lower than -55 dBm0, it is difficult to set separate limits, in view of the extremely coarse quantization of these signals. Moreover, at these levels no limits are set by Recommendation G.712 for signal to quantizing distortion ratio.

For the above reasons, no separate limits for gain variation have been considered for input levels lower than -55 dBm0.

Send side

In the study, it has been assumed that the encoder has a ± 10 % tolerance on the decision levels, and that the reconstruction levels of the decoder are at nominal values. A statistical analysis has been carried out on 100 tests with uniform probability density of the tolerance within the mentioned limits.

The histograms obtained for -50 and -55 dBm0 input level are reported in Figure A1. The results show the severity of Recommendation G.712, since its limits appear to be almost completely filled by the send side contribution alone, in the above assumptions.

On the basis of the histograms depicted in Figure A1 the following mask (send side only) has been drawn :

Level S (dBm0)	Send side gain versus level
$-50 \leq S < -10$	$\pm$ 0.3 dB
$-55 \leq S < -50$	$\pm$ 0.5 dB

Figure A1 also shows the histograms relative to the decrement of the signal-to-quantizing distortion ratio (Δ S/N). It is possible to observe that, with respect to Recommendation G.712, there are good margins in the signal-to-quantizing distortion ratio, as far the encoder precision is concerned.

The considered 10 % tolerance in the first decision levels are "reasonable" because they are congruent with good hardware implementations.

Lower values seem an excessive requirement for implementations at a reasonable cost.

Receive side

In accordance with the general trend expressed in (1) that more stringent limits should be applied to the receive side than to the send side, it has been assumed that a $\pm$5 % tolerance is on the reconstruction levels (decoder) and that the decision levels (encoder) are at nominal values.

The histograms obtained from the -50 and -55 dBm0 input signals are shown in Figure A2. On the basis of such histograms the following mask (receive side only) has been drawn :

Level S (dBm0)	Receive side gain versus level
$-50 \geq S \quad -10$	$\pm$ 0.25 dB
$-55 \geq S \quad -50$	$\pm$ 0.4 dB

Figure A2 also shows the histograms relative to the decrement of the signal-to-quantizing distortion ratio (Δ S/N).

Send and Receive side

The histograms of Figure A3 have been derived by allowing a 10 % tolerance for the send side and a 5 % tolerance for the receive side. In this way the limit of Recommendation G.712 are satisfied in practically all cases. The S/N decrements are also shown here.

Final remarks

The separate performance limits proposed, appear adequate for equipment of reasonable complexity.

It can be noted that the separate limits proposed for tests with noise signals are still more severe than those proposed for tests with sine-wave signals.

It seems reasonable to consider a statistical approach in defining separate limits. In fact, a worst-case analysis could easily lead to the practical impossibility of dividing the limits of Recommendation G.712 between send and receiv sides. As an example, Figure A4 depicts the worst cases of gain versus level for send side only ①, receive side only ② and end to end ③. To obtain the curves of Figure A4 the decision levels ± 1, ± 2, ± 3, ± 4 have been assumed to be 10 % smaller than nominal, while the reconstruction levels ± 1, ± 2, ± 3, ± 4 have been assumed 5 % greater than nominal.

Obviously this case does not satisfy Recommendation G.712 neither for end to end nor for the send side alone, the probability of this case being nevertheless negligible.

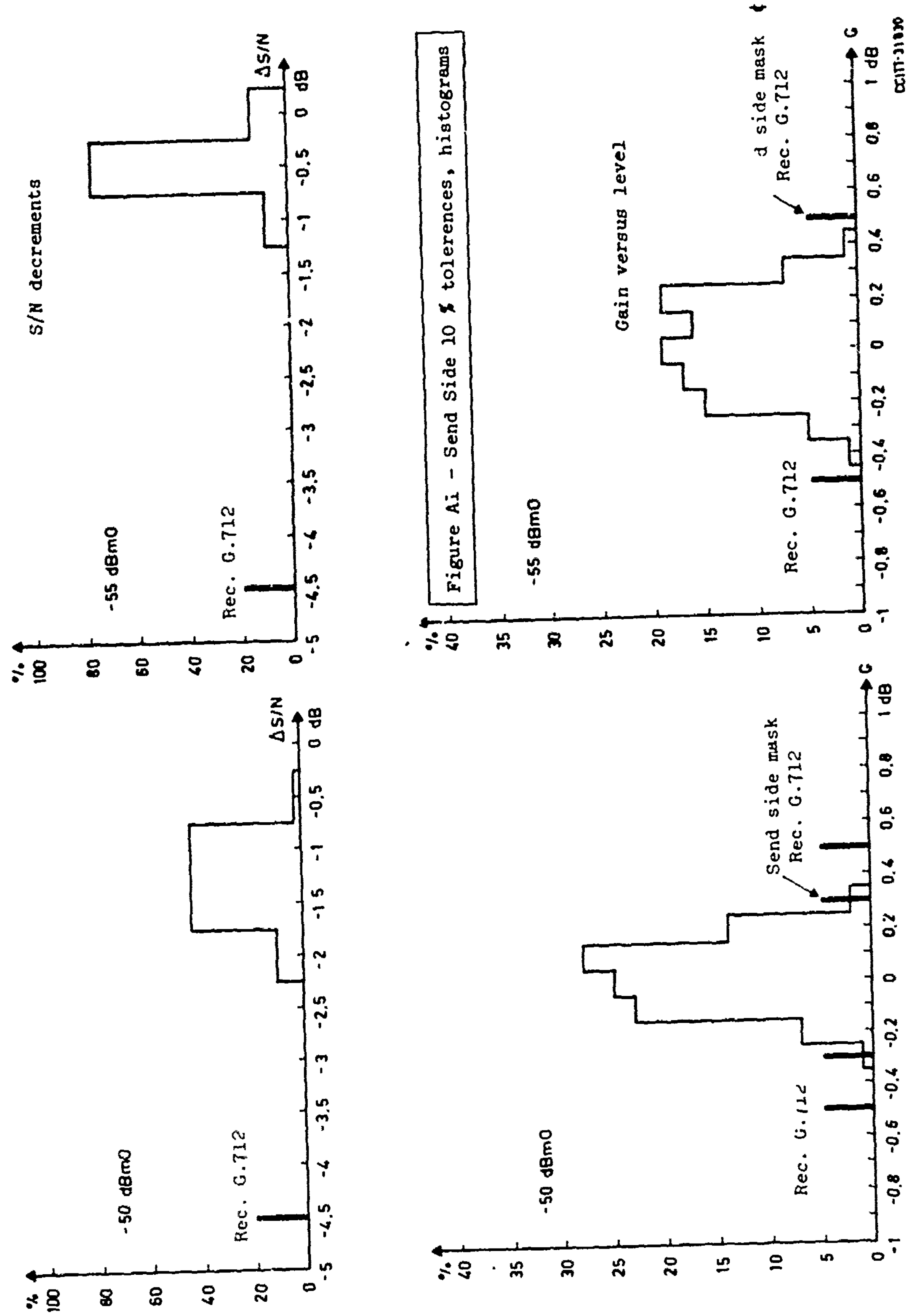


Figure A1 – Send Side 10 % tolerences, histograms

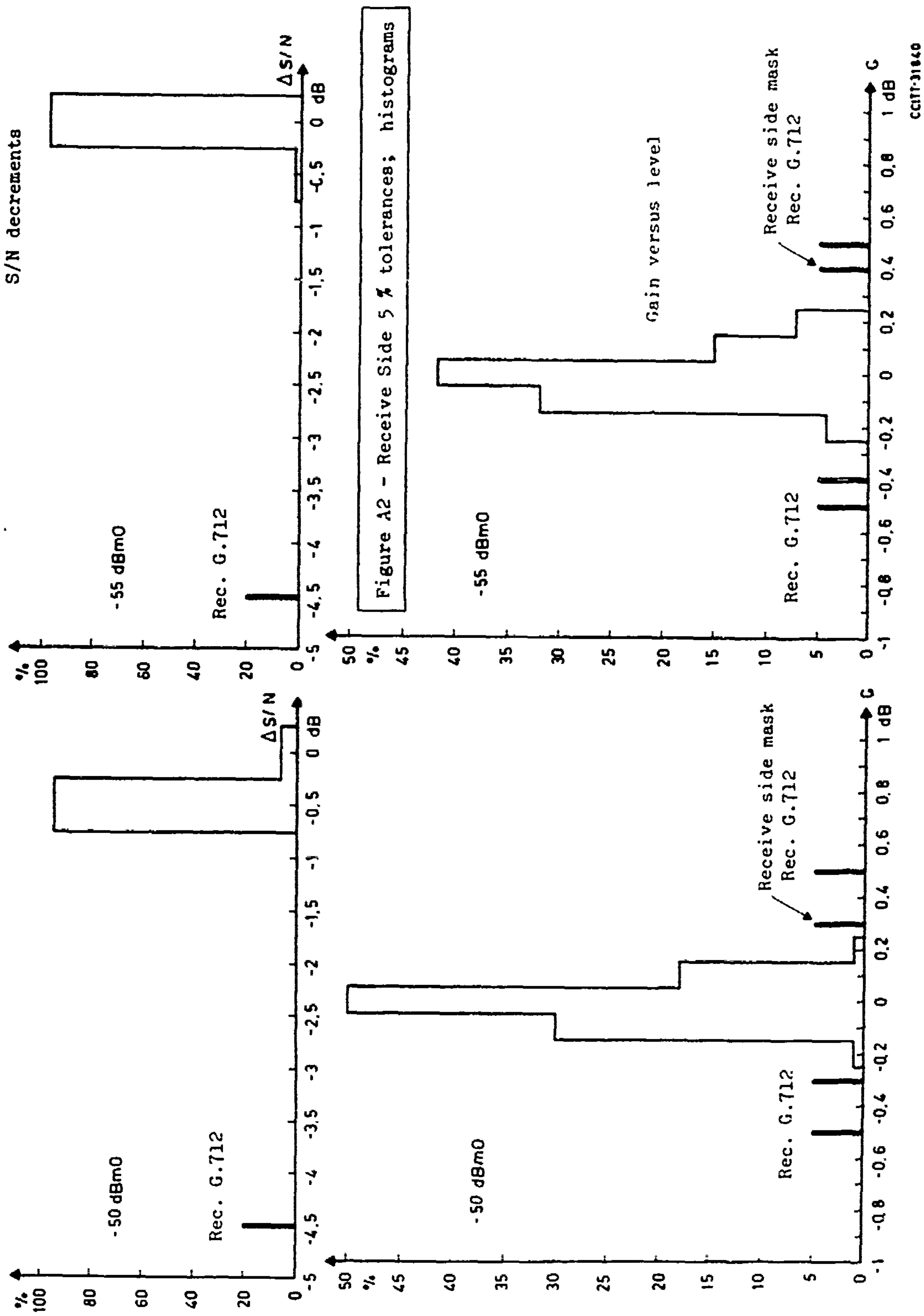


Figure A2 - Receive Side 5 % tolerances; histograms

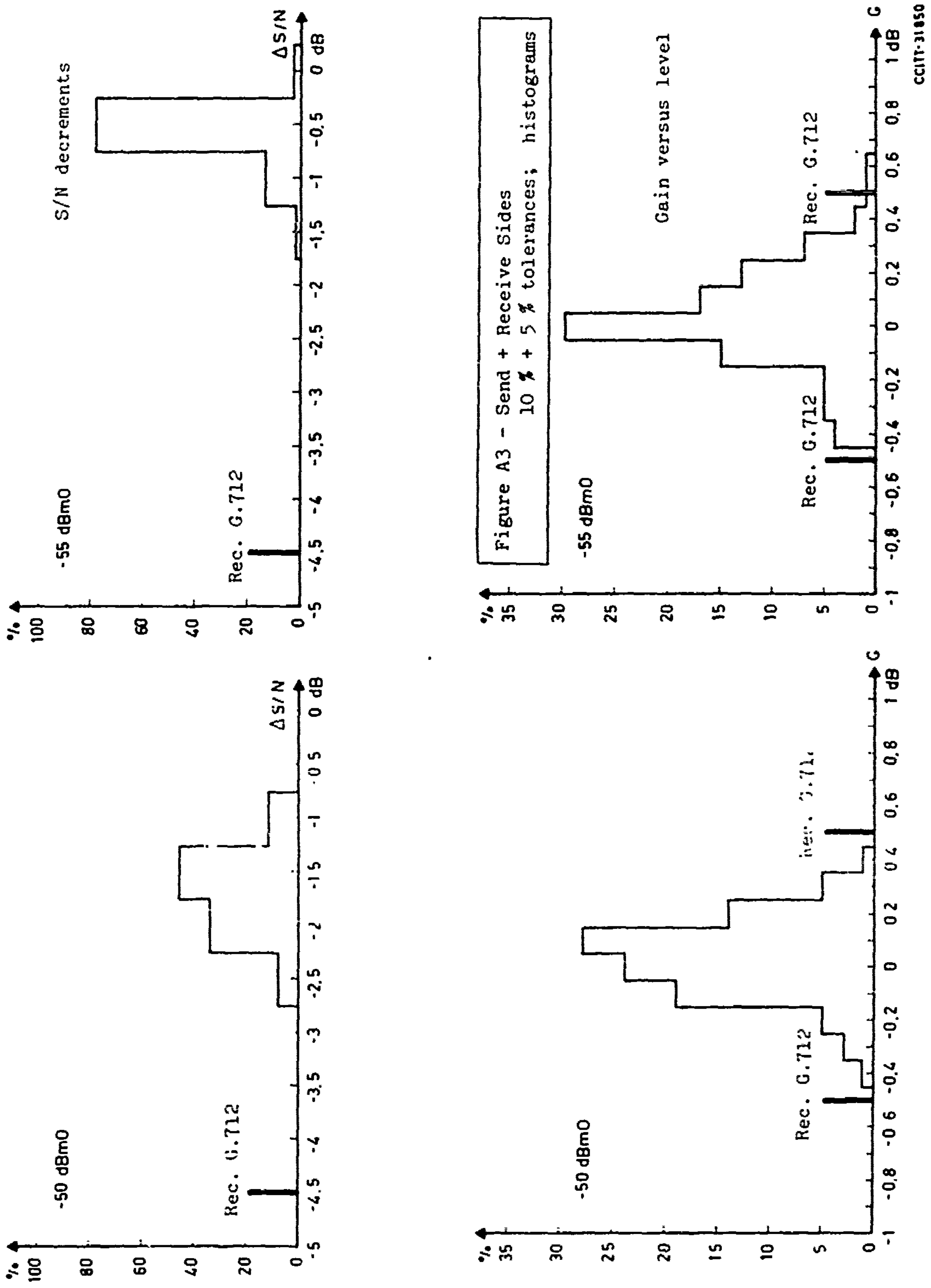


Figure A3 - Send + Receive Sides
10 % + 5 % tolerances; histograms

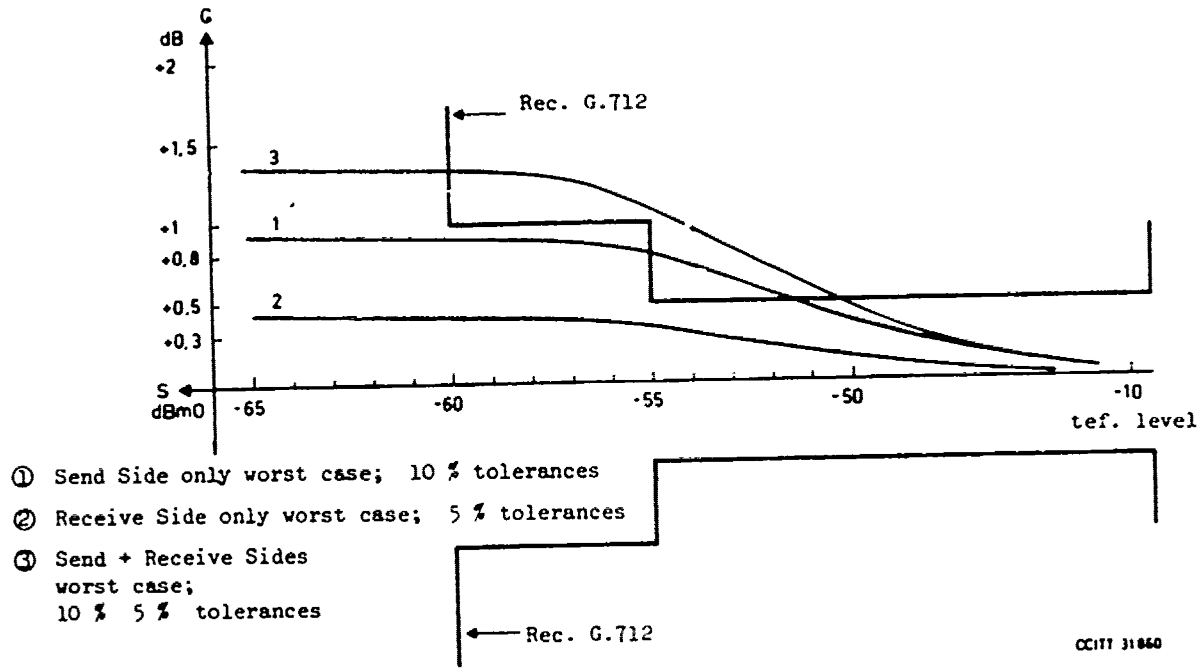


Figure A4 - Gain versus level; noise signal; worst cases.

QUESTION 17/XVIII - Characteristics of PCM multiplexing equipment and other terminal equipments for voice frequencies

(Continuation of part of Question 3/XVIII, studied in 1977-1980

considering

that recent progress in equipment design as well as maintenance philosophy studies might result in a need for modification of the present Recommendations on PCM multiplexing equipment;

that studies on encoding methods of speech and voice-band signals other than PCM and also studies on higher order PCM multiplexing equipment might result in a need for recommending new terminal equipments for voice frequencies;

a) What modifications to existing Recommendations G.731-733 and G.744 should be made ?

b) What new terminal equipment should be recommended for voice frequencies other than those recommended in Recommendations G.732, G.733 and G.744 ?

Note : Coding parameters for analogue-to-digital conversion processes, other than PCM, will be studied under Question 7/XVIII.

QUESTION 18/XVIII - Characteristics of digital multiplex equipment and multiplexing arrangements for telephony and other signals

(Continuation of part of Question 8/XVIII, studied in 1977-1980)

considering

that the introduction of large-capacity digital transmission systems as well as wideband encoders for video signals may require Recommendations on higher-order hierarchical levels and associated digital multiplexing equipment;

that octet-interleaved synchronous multiplexes may find wider or more flexible applications in digital networks;

that the progress in the digital network structure studies may result in the modification of the present Recommendations on digital exchange terminals or definition of new multiplex arrangements between exchanges;

what new or modified arrangements or characteristics for the following applications should be recommended ?

- digital multiplex equipment using justification;
- synchronous digital multiplex equipment;
- multiplexing arrangements and related characteristics to be used between digital exchanges.

In particular, the following specific points require study :

Point a) - What modification is required in the existing Recommendations relevant to digital multiplex equipment using justification techniques (G.742, G.743, etc.) ?

Point b) - What higher order or non-hierarchical digital multiplex equipment using justification should be recommended (see Question 19/XVIII) ? In particular :

1) above the fourth order at 139264 kbit/s

2) above the third order at 32064 or 44736 kbit/s

Point c) - In defining the specifications pertinent to points a) and b) above, should provisions be made to multiplex the n-th order signals directly to (n+m)-th order ($m \geq 2$) ?

Point d) - What modification is required in the existing Recommendations relevant to synchronous digital multiplex equipment and multiplex arrangement for use with digital exchanges (G.734, G.735, G.736, G.737, G.738, G.739 and G.746) ?

Point e) - What new Recommendations are required in the area of point d) above ?

Point f) - What conditions could be set concerning the use of service bits defined in the multiplex frame structures ?

Note : If interfaces should be specified for access to these bits, reference should be made to the studies undertaken under Question 3/XVIII.

QUESTION 19/XVIII - Network aspects of existing and new levels in the digital hierarchy

Considering

a) that the digital network is expected to expand both in the trunk and the local area;

b) that services other than telephony and data will have to be accommodated in the network and are expected to share the facilities to be provided;

c) that new technological development may enable efficient use to be made of existing and new transmission media;

d) that the introduction of new services may require the provision of new levels in the digital hierarchy.

1. What levels, if any, in the digital hierarchy should be defined in addition to those already recommended by CCITT ?

2. What should be the basic characteristics of multiplexes (transmission capacity, frame structure, service and maintenance facilities to be incorporated), which will serve as a basis for the subsequent studied directed as establishing recommendations for equipments ?

Note 1 : When new hierarchical levels are defined, account should be taken of (e.g. their possible use for various services).

Note 2 : The characteristics of digital line and radio sections are studied under Question 11/XVIII.

Annex

(to Question 19/XVIII)

Summary of the study performed during the 1977-1980 study period concerning digital transmission of sound programme signals

Various proposals for digital transmission of sound programme signals are summarized in Table 1. It is noted that this Table is only for information purpose.

The Appendix gives the reply to CMTT concerning digital transmission of sound programme signals.

Parameter	Italy	Switzerland	FRG	UK Note 1	France	UK Note 2	USSR	NTT	[illegible]
Nominal bandwidth (kHz)	0,04 ~ 1[illegible]	•	•	•	•	•	•	•	0,05 ~ [illegible]
Pre-emphasis characteristics	CCITT Rec. J.17	•	•	•	•	•		No pre-emphasis employed	CCITT Rec. J.17
Overload point (dBm0)	+12	•	•	•	•	•		+12	•
Sampling frequency	32 kHz ± 50 ppm	•	•	32 kHz ± 50 ppm	[illegible] kHz ± 50 ppm	•	•	•	15 kHz ± 50 ppm
Type of companding	Instantaneous	•	•	(linear)	Quasi instantaneous	•	Instantaneous	•	•
Companding characteristics	13-segment A-law	11-segment	•	(linear)	5-segment	•	7-segment	•	11-segment
Finest resolution (bit)	14	•	•	14	•	•	•	[illegible]	14
Coarsest resolution (bit)	8	9	•	–	10	•	11	10	9
Coding bit rate (kbit/s)	[illegible]	352	•	448	[illegible]	•	[illegible]	352	[illegible]
Gross bit rate (kbit/s)	[illegible]	384	•	480	384 [illegible]	•		384	192
Transmission bit rate (kbit/s)	2048	•	•	•	•	•	•	1544	2048
Number of multiplexed channels	6	5	•	4	5 6	•		4	10
Access to digital network	Synchronous	Synchronous asynchronous	Synchronous	Synchronous asynchronous	Asynchronous	•		Synchronous	Synchronous
Reference	CMTT/[illegible] (1974-1977)	CMTT/166 ([illegible]-1976)	COM XVIII-205 (1977-1980)	COM XVIII-[illegible] (1977-1980)	COM XVIII-203 (1977-1980)	-	COM XVIII-94 (1977-1980)	CMTT/14 (1974-1978) COM XVIII-157 (1977-1980)	COM XVIII-295 (1977-1980)

Table 1 - Proposals for digital sound-programme transmission

Notes to Table 1

Note 1 : Contribution submitted by the UKPC for information purposes only. The document describes a proposal that is being put forward for consideration within CMTT as a result of discussions within the European Broadcasting Union. It is aimed at overcoming the present impasse that prevents the definition of an agreed companding characteristic by suggesting that, for the time being at least, international exchange of sound programme signals could be based on 14 bit linear encoding.

Note 2 : Current UK position for a companded system.

Appendix

(to Annex to Question 19/XVIII)

Reply to CMTT concerning digital transmission of sound-programme signals

1. Introduction

In Delayed Contribution P of Study Group XVIII (draft revision of Report 647/CMTT) CMTT asks Study Group XVIII several questions relating to the digital transmission of sound programme signals. These questions are found in Annex 1 of the Document P /̲ CMTT/301(Rev.1) /̲.

The questions relate to items which are at present under study in Study Group XVIII. Consequently this document is an interim reply and further information will be forwarded when it is available.

2. Specific answers

i) Error rate, error distribution, etc.

These factors and their subdivision among various parts of the total network are currently being defined in Question 1/XVIII. Recommendation G.821 gives the results of the studies to date.

ii) Access to the network

a) The number of sound programme circuits required on international connections is generally supposed to be smaller than primary level digital path capacity which is equivalent to thirty or twenty-four voice channels. Consequently, in the case of digital sound programme transmission, it seems to be very advisable to standardize terminal station arrangement which will allow the cost-effective joint use of the primary level digital paths. For this application, the gross bit rate of a digitized 15 kHz sound programme signal of 384 kbit/s seems adequate to Study Group XVIII. The access to the primary level should be either through a new electrical interface or through an existing 2048 or 1544 kbit/s interface (cf. Recommendation G.703).

b) Where the entire primary level digital path is dedicated to a number of sound channels the access may be at the primary bit rate. The gross bit rate of a digitized 15 kHz sound-programme channel may be 384 kbit/s (5 channels in 2048 kbit/s, 4 channels in 1544 kbit/s). However, in the case of 2048 kbit/s hierarchy, it could be reduced below 384 kbit/s in order to provide 6 channels on one 2048A kbit/s digital path. In the latter case it could be necessary to define a frame structure of the 2048 kbit/s signal different from that recommended in Recommendation G.732.

iii) Synchronous or asynchronous access

a) In case ii (a) :

Where the primary multiplex level is shared with other services such as telephony or data, the access to the primary level may be synchronous with the multiplex. This will lead to low-cost, single-channel basis digital interconnection including switching or multi-junction between two or more primary digital paths. Necessary timing signals to synchronize sound programme codecs could be provided in the form of the centralized-clock, codirectional or contradictional interface as is exemplified by Recommendation G.703.

In the case where the introduction of synchronization in digital networks is unlikely to be geared to the needs of digital sound programme transmission and taking into account the evolution towards a synchronous network, the access to the primary level should be in such a way as to allow asynchronous operation using justification technique.

The content of a digital signal at 384 kbit/s should be defined, which includes a small amount of spare capacity for housekeeping functions including the justification information (not more than a few kbit/s).

Study Group XVIII shall define a multiplexing method for synchronous access (which will not make use of this spare capacity for justification) and for asynchronous access (which does make use of the spare capacity).

The system concept envisaged is given in Figure 1. Study Group XVIII will decide later if a new digital interface should be specified or not.

It is expected, that the clock-frequency inaccurracies of the encoder and of the parts of the network, in the case of asnchronous access, will not be more than ± 50 ppm each.

b) in case ii (b) :

Since a complete primary level is used, the access need not be synchronous with the rest of the network. If switching in the digital mode of sound programme channels is envisaged, slip will occur unless the total network incolved is synchronized.

iv) Tariff principles

Tariff principles are not the responsibility of Study Group XVIII.

v) Aspects of multiplexing and network synchronization

CCITT has recommended in Recommendation G.811 that the international digital links should be operated in a plesiochronous manner with reference clock of very high accuracy (10^{-11}). This implies that the national networks are either fully synchronous or plesiochronous with the same accuracy.

It is pointed out that a slip rate of 1 slip in 70 days per plesiochronous interconnection is the resulting theoretical slip rate, taking into account clock accuracies according to Recommendation G.811 only. However, account should also be taken of practical network characteristics encountered under normal operating conditions even where synchronized national networks remain synchronized. As a guide, draft Recommendation G.822 for a 64 kbit/s end-to-end connection (switched or permanent), mentions suggested value of 1 slip in 5 hours.

Digital sound programme channel interconnection can be realized without any frequency adaptation if the terminal stations belong to the same synchronous network. Simple frequency adaptation by means of slip technique is permitted if the terminal stations belong to different synchronous networks with the clock-accuracy as specified in Recommendation G.811.

However, in the evolution towards this situation the terminal station will likely be operated at lower clock accuracies as specified by Recommendations G.732 and G.733 (5×10^{-5}). In this case the sound programme signal should be justified at all asynchronous interconnecting points, or the interconnection should be at analogue interfaces.

It is noted that the above synchronous interconnection applies without any restriction as long as the Telecommunication Administrations are responsible for terminal station including encoding. In the event that encoding should become the responsibility of the Broadcasting Organizations, it is only applicable if their clocks are in synchronism with that in the network of the Administration concerned.

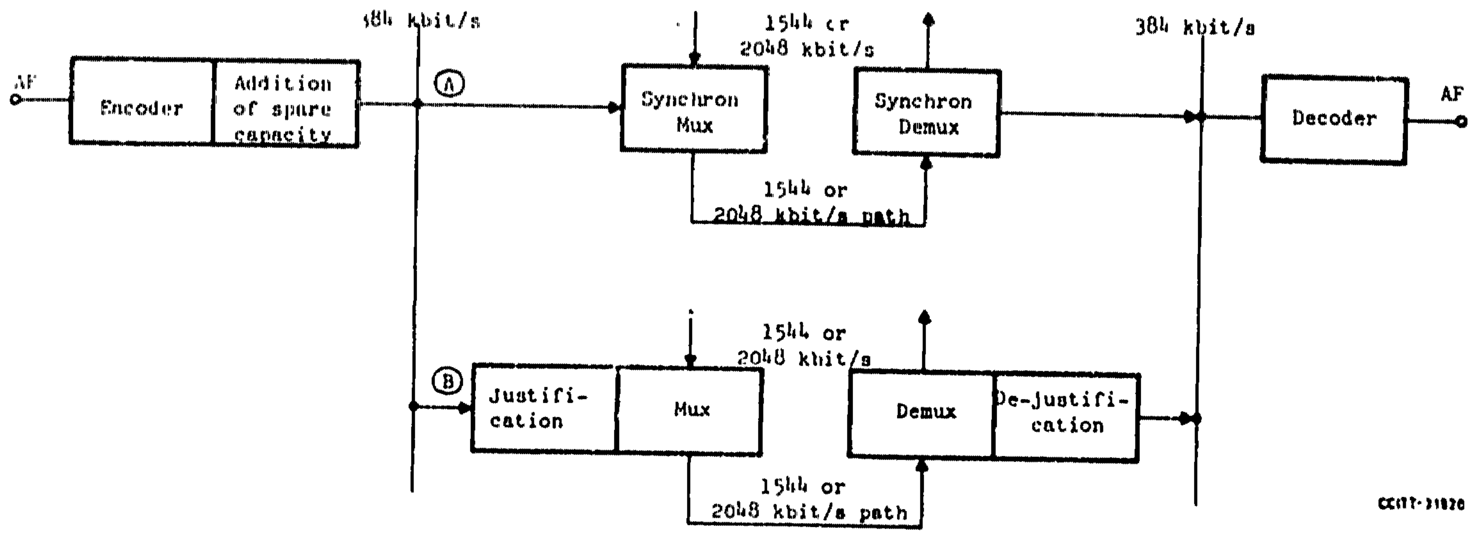


Figure 1 - System concept envisaged for case ii (a)

(A) Synchronous access to 1544 or 2048 kbit/s digital path

(B) Asynchronous access (to be used where it is necessary)

Note 1 : The timing of the encoder is not shown in this figure.

Note 2 : There is no interface at 384 kbit/s standardized. In this figure, only the capacity allocated to one 15 kHz channel is indicated.

A N N E X

(to the list of Questions)

STUDY OF THE INTEGRATED SERVICES DIGITAL NETWORK (ISDN)

Examination of the Questions drafted by Study Groups III, IV, VII, XI, XVII and XVIII reveals that each Group intends to study various aspects of the ISDN. In order to avoid overlapping and possibly conflicting results the VIIth CCITT Plenary Assembly agreed that the areas of responsibility for the study of ISDN should be assigned as follows :

		Assign to :
1.	Services and facilities interpretation and coordination (taking into account the requirements identified by Study Groups I, II, III and VII).	XVIII
2.	General ISDN aspects and guidelines, quality of service, numbering, performance targets, maintenance principles and miscellaneous subjects not more specifically identified (taking into account the requirements of Study Groups I, II, IV, VII, XI, XVII, and CMBD). Note : It is considered that Items 1 and 2 above are of high priority.	XVIII
3.	Digital transmission standards and performance (local and inter-exchange). The study of hypothetical reference connections is in the competence of Study Group XVIII, the study of hypothetical reference digital paths is in the competence of the specialized Study Groups of CCITT and CCIR, the study of reliability and availability is to be coordinated by CMBD. Note : Also of interest to CCIR.	XV/XVIII
4.	Switching aspects and parameters (taking into account the requirements identified by Study Groups VII, XVII and XVIII). Note : In the case of mixed mode switches (e.g. ISDN circuit and packet) other Study Groups will also be consulted.	XI
5.	Inter-exchange signalling system (Message Transfer Part (MTP) and appropriate User Part(s)) (taking into account the requirements identified by Study Groups VII and IX).	XI
6.	Subscriber-exchange signalling system (taking into account the requirements identified by Study Groups I, II, VII and XVII and coordinated by XVIII - see Item 2).	XI

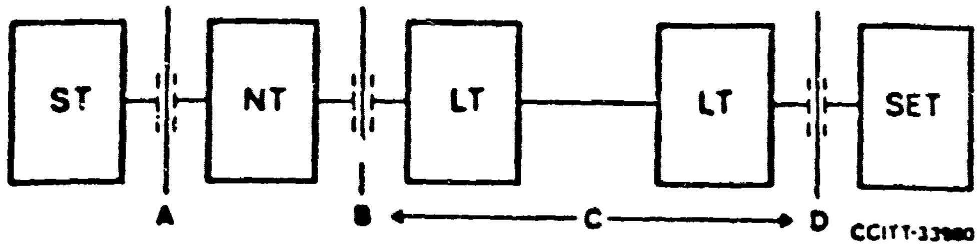
		Assign to :
7.	Subscriber-network interface	
	i) Interface B	XI
	ii) Interface A - Voice services	XI
	iii) Interface A - Non-voice services	VII/XVII
	iv) Interface A - Alternate voice/data	VII/XI/XVII

Note : Close collaboration between Study Groups VII, XVII and XI will be required to ensure compatibility between i), ii), iii), iv) and the subscriber signalling system identified in 6.

8.	Interworking (inter-service and inter-network)	
	i) Data	VII
	ii) Telex	IX
	iii) Telephone	XI
	iv) Data over the telephone network	I/II/XVII
	v) Teletex	I/VIII
	vi) Facsimile	I/VIII

Note : Collaboration between the Study Groups referred to above will be required to ensure compatibility in the carriage of the various services on ISDN and other networks.

9.	Digital telephone instrument	XII
10.	Tariff aspects	III



LEGEND

ST - Subscriber terminals

NT - Network termination

LT - Line terminal

SET - Subscriber line exchange termination

Functional interfaces A, B, D

C Subscriber line transmission

Figure 1 - Possible functional interfaces*) in digital local access

---

*) "Interface" - "a concept involving specification of the interconnection ...." page 89, Orange Book, "Definitions"